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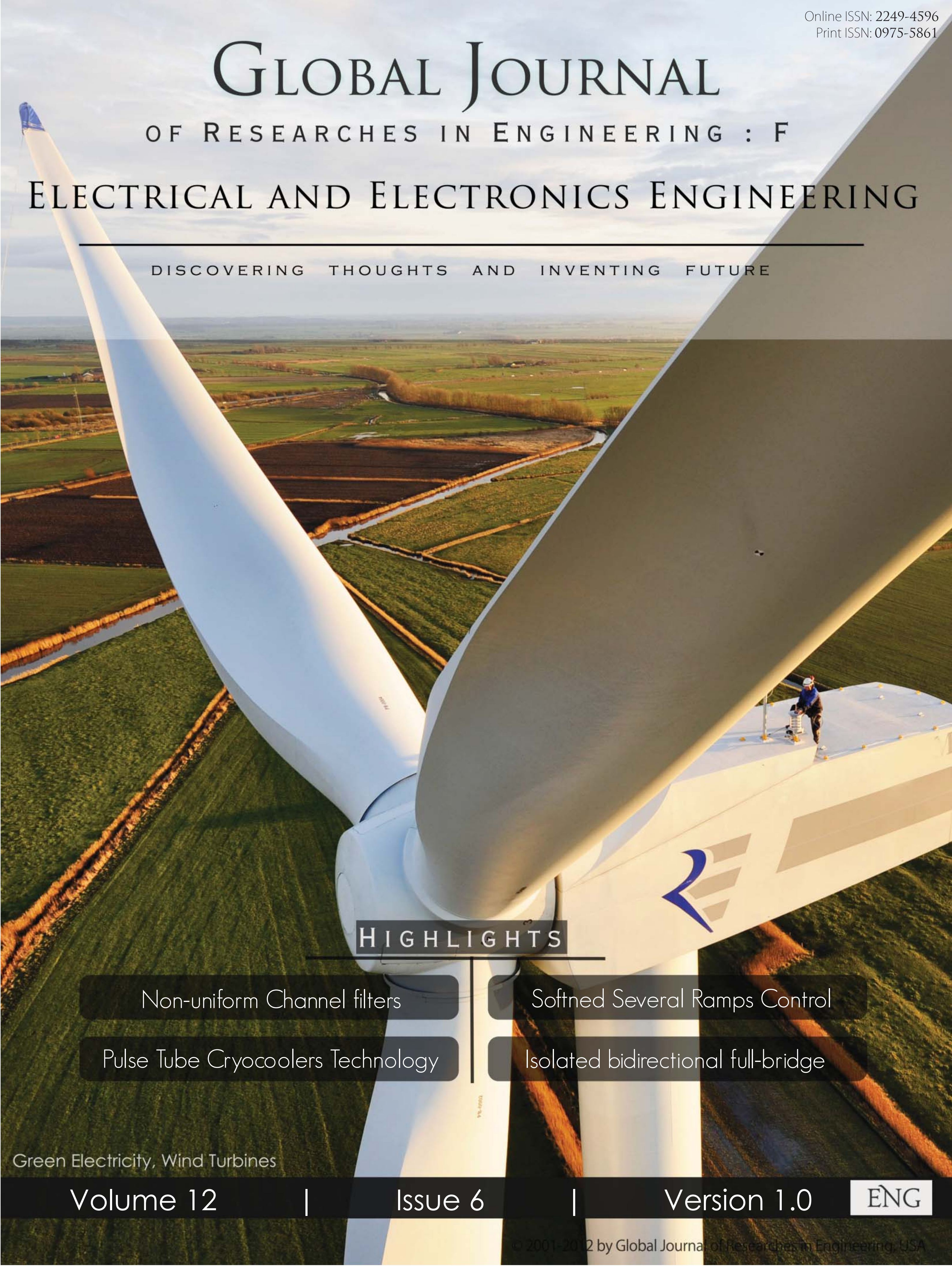

Highlights

- Non-uniform Channel filters
- Softned Several Ramps Control
- Pulse Tube Cryocoolers Technology
- Isolated bidirectional full-bridge

Green Electricity, Wind Turbines

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Reduction of The Chattering by Using Softned Several Ramps Control Law in the Field-Oriented Control Process of an Induction Motor

By A.Yahi, L. Barazane & M.S.Boucherit
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Abstract - The aim of this paper is to propose a new control law for induction motor squirrel cage that is deemed by its strength, high torque mass, and its relatively low cost ... etc, meanwhile, it benefited from the support of industry since its invention. Despite these advantages, these actuators are complex dynamic systems that exhibit a strong non-linearity, which makes them difficult to control. So, the variable structure control is a solution for this problem because it is robust control despite the uncertainty on the model, disturbances and parametric variations. In practical applications, the main disadvantage associated with the command is the phenomenon of chattering. In order to minimize the amplitude of oscillations we propose a new technique of variable structure control which is softened several ramps. A simulation results revealed some very interesting features.

Keywords : Induction motor, field-oriented control, sliding mode control.

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A.Yahi [α], L. Barazane [σ] & M.S.Boucherit [ρ]

Abstract - The aim of this paper is to propose a new control law for induction motor squirrel cage that is deemed by its strength, high torque mass, and its relatively low cost ... etc, meanwhile, it benefited from the support of industry since its invention. Despite these advantages, these actuators are complex dynamic systems that exhibit a strong non-linearity, which makes them difficult to control. So, the variable structure control is a solution for this problem because it is robust control despite the uncertainty on the model, disturbances and parametric variations. In practical applications, the main disadvantage associated with the command is the phenomenon of chattering. In order to minimize the amplitude of oscillations we propose a new technique of variable structure control which is softened several ramps. A simulation results revealed some very interesting features.

Keywords : Induction motor, field-oriented control, sliding mode control.

I. Introduction

For many years, the DC machine has taken a special place and distinguished in various industrial applications. This thanks to its simplicity of control due to the natural decoupling between torque and flux, and also its unmatched dynamic performance that let such a machine used in different speeds in various processes. However, the sparks are due to brush-collector contact and the volume of the latter make use of this type of machine useless [1-2].

On the other hand, the asynchronous squirrel-cage which is famous for its strength, high torque mass, and its relatively low cost etc ...., Meanwhile, has enjoyed the favor of the industry since its Nicola Tesla invention by the end of last century [3]. However, the dynamics of this type of machine is found to be non-linear, multi-variable and strongly coupled resulting in rather poor performance in operation with the control V / F = constant. This is unlike the case of the DC machine which is considered as an optimum in point of view simplicity of control compare to the other machines in general that we try to find. This is due to the simplifications offered by the system brushes-collector [2-3].

On the other hand, thanks to the evolution witnessed in the field of power electronics components and the different control techniques applied to the induction machines with squirrel-cage, most of the recent industrial application and control motor drivers are based on induction motor and make such processes as performed as the DC machine. In this sense, the first technique that was used is called "vector control". This technique allows obtaining a decoupled dynamic model similar to the model of the machine to separate DC excitation [4-5-6].

In practice, the use of conventional correction schemes for vector control of induction motor cage where the model is nonlinear and variable parameters seems not so non-robust and efficient [5-7]. Indeed, it is known that such control is very sensitive to any changes in the motor rotor resistance, which is why research has for decades referred to other types of controllers more robust among others: those to changing patterns that were used in this paper.

This work is structured in three stages: In the first step, we present the model of the inverter three-level PWM. The second stage will be devoted to the design of indirect vector control scheme based on a cruise of classic IP. So, in this context and to do this, the basic principles of this type of technical orientation of rotor flux will be presented.

In the third step, the basic concepts of control structures and variables will be used for the detailed design of a sliding mode control with the prior the first discontinuous control law which is : the boundary layer. Thereafter and to reduce the chattering phenomenon, we propose a new law based on the command softened several ramps.

Finally, numerical simulations and comparisons of results will be presented with the aim of validating the proposed approach.

II. Three-Level Npc Vsi Model

The three phase's three-level NPC VSI structure is illustrated in figure.1. This converter is constituted by three arms. Every arm has four bi-directional switches and two clamping diodes.

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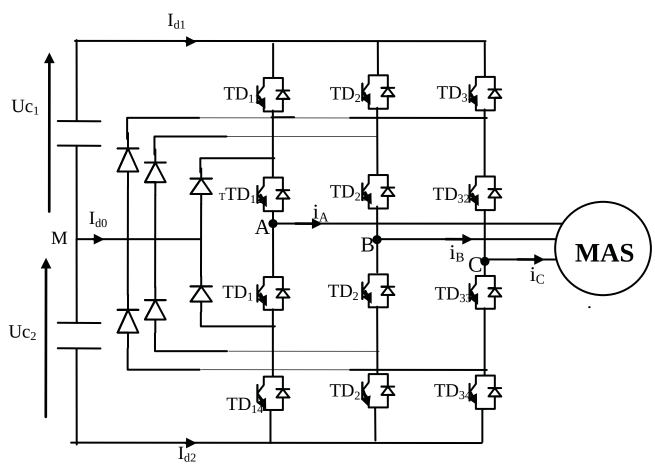

Fig.1 : Three level tension inverter NPC VSI structure.

In controlling mode, the optimal complementary law for the three-level NPC VSI is given the following equation:

$$B_{i1} = \overline{B_{i4}} \qquad B_{i2} = \overline{B_{i3}} \tag{1}$$

With:

B_{is} is the control signal of the semi-conductor TD_{is} of the inverter.

The input voltage of the rectifier, relatively of the neutral point N, is represented by the following system:

$$\begin{pmatrix} V_A \\ V_B \\ V_C \end{pmatrix} = 1/3 \begin{pmatrix} 2 & -1 & -1 \\ -1 & 2 & -1 \\ -1 & -1 & 2 \end{pmatrix} \begin{pmatrix} f_{11}^b - f_{10}^b \\ f_{21}^b - f_{20}^b \\ f_{31}^b - f_{30}^b \end{pmatrix} \cdot \begin{pmatrix} U_{c1} \\ U_{c2} \end{pmatrix} \tag{2}$$

Where: f_{i1}^b and f_{i0}^b are respectively the upper and lower half-arm connection functions of the inverter.

$$f_{i1}^b = f_{i1} . f_{i2} \qquad f_{i0}^b = f_{i3} . f_{i4} \tag{3}$$

With *i*=1, 2 and 3 is the number of phase.

The performances of this inverter are presented in figure.2.

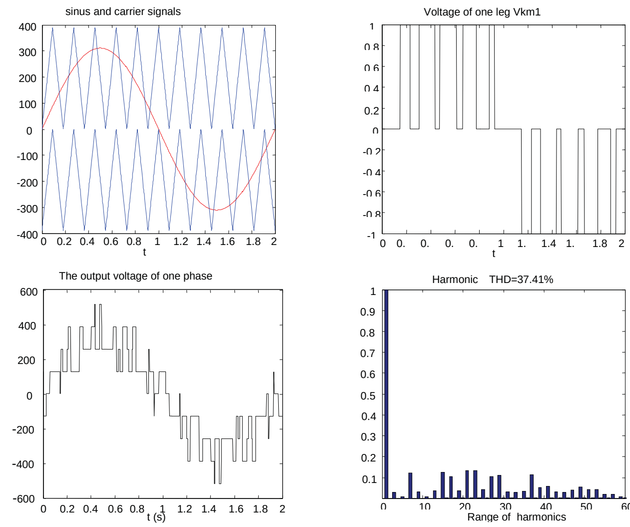

Fig.2 : Characteristics of the simple output voltage Va of the three phases three levels of cells overlapped VSI and its spectrum with r=0.8 ; m=11.

with:

1) modulation index m,
2) and the modulation rate r.

III. Control of an Induction Motor

a) Model of the induction motor

The equations of the voltage PWM source inverter fed induction motor with current control, in the synchronous reference frame (d–q), using rotor fluxes as state variables are given by [8]:

$$v_{ds} = \sigma\, L_s \frac{d\,\iota_{ds}}{d\,t} + R_s\,\iota_{ds} - \sigma\, L_s \omega_s \iota_{qs} - \frac{L_m}{T_r}\,\omega_r \Phi_{qr} \tag{4.a}$$

$$v_{qs} = \sigma\, L_s \frac{d\,\iota_{qs}}{d\,t} + R_s\,\iota_{qs} + \sigma\, L_s \omega_s \iota_{ds} + \frac{L_m}{T_r}\,\omega_r \Phi_{dr}$$

$$\frac{d\Phi_{dr}}{dt} = \frac{-1}{T_r}(\Phi_{dr} - L_m \cdot \iota_{ds}) + \omega_{sl} \cdot \Phi_{qr}$$

$$\frac{d\Phi_{qr}}{dt} = \frac{-1}{T_r}(\Phi_{qr} - L_m \cdot \iota_{qs}) + \omega_{sl} \cdot \Phi_{dr} \tag{4.b}$$

$$\frac{d\Omega}{dt} = \frac{1}{J} \cdot (T_{em} - T_L - f_r \cdot \Omega)$$

$$T_{em} = \frac{3}{2} \frac{pL_m}{L_r} \cdot (\Phi_{dr} \cdot \iota_{qs} - \Phi_{qr} \cdot \iota_{ds})$$

For a rotor-flux orientation, the regulator imposes the orientation of the rotor flux (Φ_r) with respect to the *d*-axis, giving $\Phi_r = \Phi_{dr}$ and $\Phi_{qr} = 0$. Substituting these relations in (5), leads to the field-oriented model of the motor which is given by the following equation system:

$$v_{ds} = \sigma\, L_s \frac{d\,\iota_{ds}}{d\,t} + R_s\,\iota_{ds} - \sigma\, L_s \omega_s \iota_{qs} - \frac{L_m}{T_r}\,\omega_r \Phi_r$$

$$v_{qs} = \sigma L_s \frac{d\,\iota_{qs}}{d\,t} + R_s\,\iota_{qs} + \sigma L_s \omega_s \iota_{ds} + \frac{L_m}{T_r}\,\omega_r \Phi_r$$

$$\frac{d\Phi_r^{\,*}}{dt} + \frac{1}{T_r}\cdot\Phi_r^{\,*} = \frac{L_m}{T_r}\cdot\iota_{ds}^{\,*}$$

$$\omega_{sl}^{\,*} = \frac{L_m\cdot\iota_{qs}^{\,*}}{T_r\cdot\Phi_r^{\,*}} \qquad (5)$$

$$T_{em}^{\,*} = \frac{3}{2}\frac{pL_m}{L_r}\Phi_r^{\,*}\cdot\iota_{qs}^{\,*}$$

The field-oriented controller is based on the inversion of the above equation system. The command variables (i_{ds}^{*}, i_{qs}^{*}, v_{ds}^{*}, v_{qs}^{*}) are generated here respectively by regulators as it is shown in figure.3.

The rotor flux is estimated by means of stator current and speed measurements (direct method) as follows:

$$\frac{d\,\iota_{ds}}{dt} = \frac{1}{\sigma L_s}\left[v_{ds} - \left(R_s + \left(\frac{L_m}{T_r}\right)^2\right)\cdot\iota_{ds} + \right.$$
$$\left. + \sigma L_s \omega_s \iota_{qs} + \frac{L_m}{T_r}\omega_r \Phi_r\right] \qquad (6)$$

$$\frac{d\,\iota_{qs}}{dt} = \frac{1}{\sigma L_s}\left[v_{qs} - \left(R_s + \left(\frac{L_m}{T_r}\right)^2\right)\cdot\iota_{qs}\right.$$
$$\left. - \sigma L_s \omega_s \iota_{ds} + \frac{L_m}{T_r}\omega_r \Phi_r\right]$$

The corresponding position is given by:

$$\theta_e = \int (p\cdot\Omega + \omega_{sl}^{\,*})\; dt \qquad (7)$$

IV. Variables Structures Control

The basic principle of the variables structures control also called the sliding mode control consists in moving the state trajectory of the system toward a predetermined surface called sliding or switching surface and in maintaining it around this latter with an appropriate switching logic. This is similar to a feed-forward controller that provides the control that should be applied to track a desired trajectory, which is in this case, the user-defined sliding surface itself. So, the design of a sliding mode controller has two steps, namely, the definition of

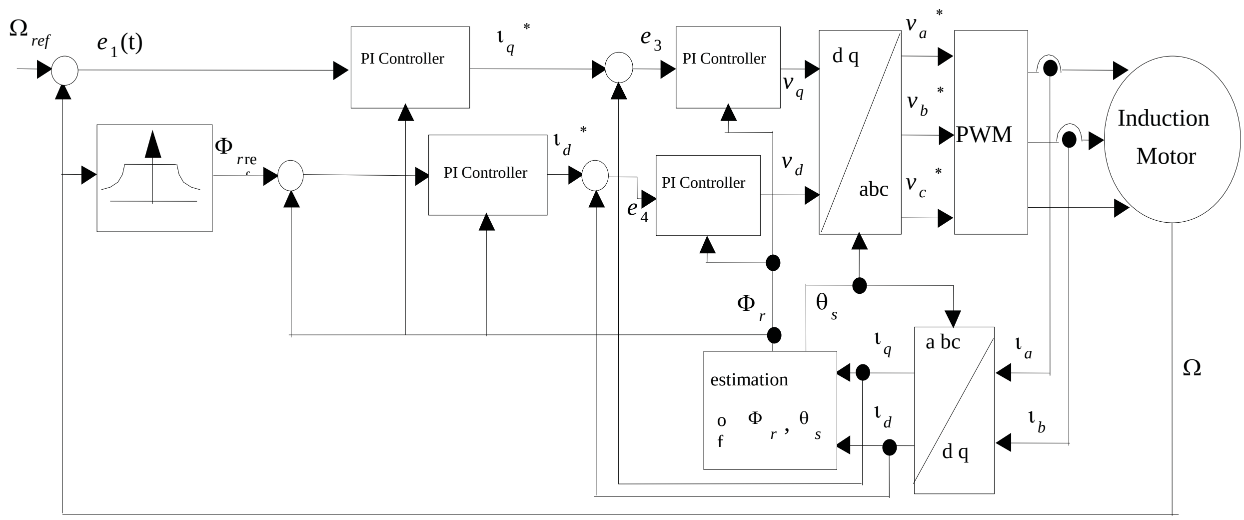

Fig. 3 : Vector control scheme of an induction motor.

the adequate switching surface $S(\cdot)$ and the development of the control law or the switching logic U.

Concerning the development of the switching logic, it is divided into two parts, the equivalent control U_{eq} and the attractivity or reachability control U_n. The equivalent control is determined off-line with a model that represents the plant as accurately as possible and calculated by imposing the derivative of sliding mode surface equal to zero. If the plant is exactly identical to the model used for determining U_{eq} and there are no disturbances, there would be no need to apply an additional control U_n. However, in practice there will be discrepancy between the model and the actual system control. Therefore, the control component U_n is necessary and it will always guarantee that the state is attracted to the switching surface by satisfying the following attractivity condition.

$$S(\cdot)\cdot\dot{S}(\cdot)<0 \tag{8}$$

Therefore, the basic switching law is of the form:

$$U = U_{eq} + U_n \tag{9}$$

With: $U_n = -M(\cdot)\cdot \mathrm{Sgn}(S(\cdot))$

$M(S)$: the magnitude of the attractivity control law U_n,

Sgn: the sign function.

In a conventional the variables structures control (VSC), the reachability control generates a high control activity as it depends on the magnitude $M(\cdot)$ since it was first taken as constant, a relay function, which is very harmful to the actuators and may excite the unmodeled dynamics of the system. This is known as a chattering phenomenon. Ideally, to reach the sliding surface, this phenomenon should be eliminated [9-10]. However, in practice, chattering can only be reduced.

To reduce chattering was to introduce a boundary layer around the sliding surface and to use a smooth function to replace the discontinuous part of the control action as follows [8-10]:

$$U_n = \begin{cases} \dfrac{K}{\varepsilon}\cdot S(x) & \text{if } |S(x)| < \varepsilon \\ K \cdot Sgn(S(x)) & \text{if } |S(x)| > \varepsilon \end{cases} \tag{10}$$

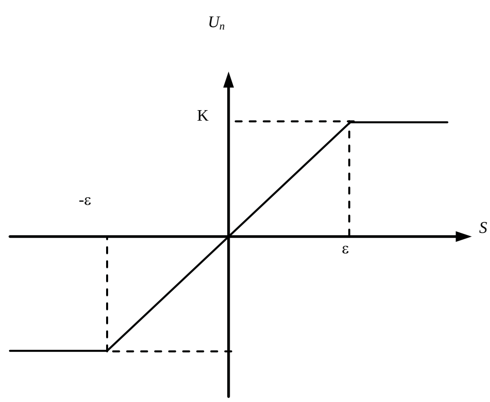

Fig.4 : Sliding mode with boundary layer and the modified switching law.

The constant K is linked to the speed of convergence towards the sliding surface of the process the reaching mode). Compromise must be made when choosing this constant, since if K is very small the time response is important, whereas when K is too big, the chattering phenomenon increases so, in this paper, we proposed an approach which is able to reduce this latter and will be described in the following section.

V. Control Softened Several Ramps

In order to further reduce the chattering, the idea consists to approach gradually the control state of the sliding mode surface. So, we propose to use a softened several ramp discontinuous control law given in eq.11. Thus, this control law must be based on the approach of the state toward the surface in the region that borders the latter, and that following several slopes with a pass through the origin in the $(S(x), Un)$ in order to avoid a discontinuity of operation as it is shown in figure.5 [11].

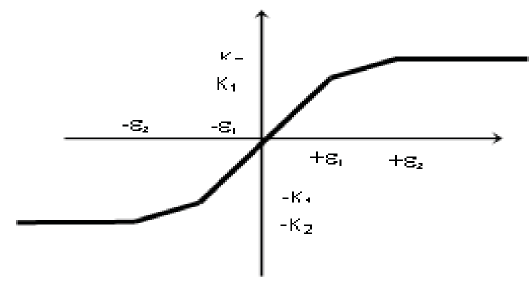

Fig.5 : Sign function control with several ramps.

The expression that corresponds to it is given by:

$$U_n = M(S).sign(s) \tag{11}$$

with:

$$M(S) = \begin{cases} +K2 & S(x) > +\varepsilon 2 \\ ((K_2-K_1)/(\varepsilon_2-\varepsilon_1)).\ S(x) & \varepsilon 1 < |S(x)| < + \\ (K_1/\varepsilon_1).S(x) & -\varepsilon_1 < |S(x)| < \varepsilon_1 \\ ((K_1-K2)_1/(\varepsilon_1-\varepsilon_2)).\ S(x) & -\varepsilon 2 < |S(x)| < -\varepsilon 1 \\ -K_2 & |S(x)| < -\varepsilon_2 \end{cases}$$

To satisfy the stability condition of the system, the gains K_1 and K_2 should be taken positive by selecting the appropriate values.

VI. Conception of the Speed Variables Structures Controller of the Cascade Scheme

In this paper, only the PI speed controller of the figure.3 is replaced by variables structures one. Its corresponding parameters are defined as follows:

a) Design of the switching surface

The sliding mode surface is defined as:

$$S(\Omega) = (\Omega_{ref} - \Omega) + m_1 \int (\Omega_{ref} - \Omega)dt \tag{12}$$

With: Ω_{ref} being respectively the reference variable of the rotor speed.

b) Development of the control law

By using the equation system (5), the speed regulator control laws are obtained as follows:

$$S(\Omega)\cdot\dot{S}(\Omega)<0 \;\Rightarrow\; \iota_{qs}=\iota_{qseq}+\iota_{qsn} \tag{13}$$

With:

$$\iota_{qseq}=\frac{J\cdot\dfrac{d\Omega_{ref}}{dt}+T_L}{\dfrac{3}{2}\,p\,\dfrac{L_m}{L_r}\cdot\Phi_r}$$

Concerning the discontinuous control law ι_{qsn} it is considered as it was defined in eq.10 and eq.11 respectively.

VII. Validation of the Cascade Sliding Mode Controllers

In order to verify the effectiveness of the softened several ramps control law (eq.11) compared to the boundary layer one (eq. 10), a simulation of the dynamic of the process is done by considering the following tests:

The first test concerns a no-load starting of the motor with a reference speed Ω_{ref} = 100 rad/sec. Then a torque load $(T_L = 10\ \mathrm{Nm})$ is applied at t = 0.5 sec. The results are shown in “fig.6 and 7”.

Note that the parameters of the induction motor used are given in Appendix.

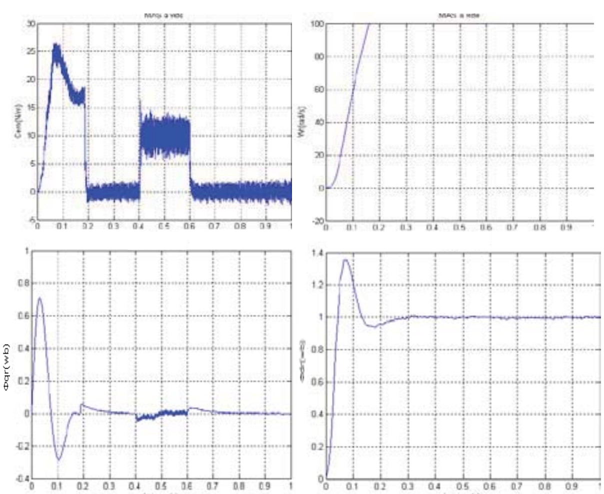

Fig. 6 : Dynamics simulation of induction machine with a perturbation of a load torque of 10 Nm with a boundary layer

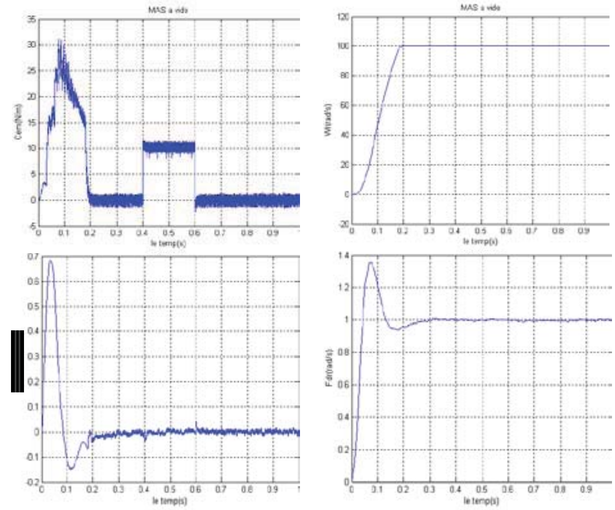

Fig 7 : Dynamics simulation of induction machine with a perturbation of a load torque of 10 Nm with softened several ramps.

It is noticed that the speed regulation is obtained using such a controller is spite of the presence of stern disturbances such as step change of the load torque.

The waveforms depicted in figure.7 show that the ideal field-oriented control is established by setting the flux responses Φ_{qr} = 0, Φ_{dr} = 1 wb, despite the load variations.

The step changes in the load torque and the reverse of speed response cause step changes in the torque response without any effects on the fluxes responses, which are maintained constant, due to the decoupled control system between the torque and the rotor flux. Thus, the aim of the field-oriented control is achieved, and the introduction of perturbations is immediately rejected by the control system. The chattering phenomenon appears in the torque response due to the discontinuous characteristic of the controller and a marked reduction of such a phenomenon with the second discontinuous control law is noticed. So, we can see that our objective is attempted with great success.

On the other hand, in order to test the robustness of this control towards the parameters variation especially the rotor’s resistance which could leads to the loose of the vector control of the motor, we made a variation of rotor resistance by 50% from its nominal value. The results are given in figures (8)

From the obtained waveforms, it is clearly shown that the robustness of our system toward parameter’s variations and external perturbation is ensured in all the sliding mode control processes which confirms the ability of these kind of structures variables control techniques to ensure the tracking and the robustness as it is cited by numerous other works.

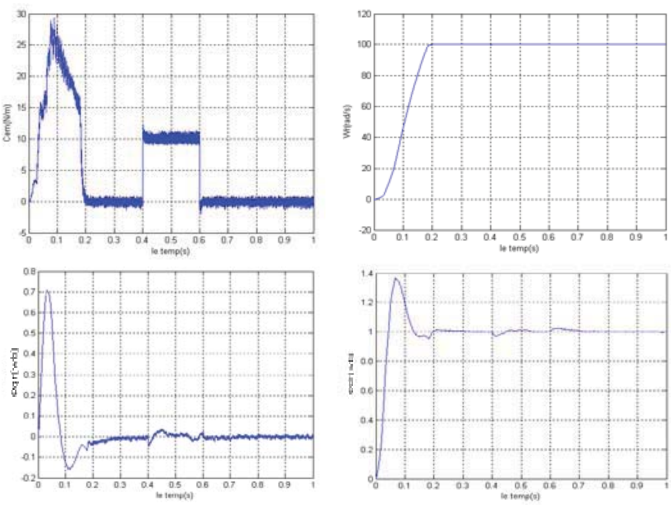

Fig .8 : Dynamics simulation of induction machine with a perturbation of a load torque of 10 Nm with variation Rr 50%.

VIII. Conclusion

The proposed approach has revealed very interesting features. In fact, the combination of the nonlinear control with the field oriented control maintains an effective decoupling between speed and flux for the whole range of speed which allows to obtain high dynamic performances for constant flux operation similar to that of dc motors. Further, these high performances are maintained above the nominal speed for the constant power operation, which is not the case in the conventional field oriented control. The addition of the sliding mode controllers has improved the robustness towards internal parameter variations, modelling uncertainties and external disturbances.

IX. Appendix: Machine Parameters

Squirrel-cage induction motor of 1.5 Kw, 220 V, 2 poles, 1420 tr/min, 50 Hz.

$R_s = 4.85\ \Omega$; $R_r = 3.805\ \Omega$; $L_s= 0.274$ H ; $L_r = 0.274$ H; M = 0.258 H ; J = 0.031 Kg.m^2 $f= 0.00114$ Nms

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Design of Low Power and Low Complexity Multiplier-less Reconfigurable Non-uniform Channel filters using Genetic Algorithm

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DESIGNOFLOWPOWERANDLOWCOMPLEXITYMULTIPLIER-LESSRECONFIGURABLENON-UNIFORMCHANNELFILTERSUSINGGENETICALGORITHM

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Design of Low Power and Low Complexity Multiplier-less Reconfigurable Non-uniform Channel filters using Genetic Algorithm

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Abstract - The key requirements of the channel filter in a software defined radio receiver are low power, low complexity and reconfigurability of the architecture used. An architecture based on frequency response masking (FRM) technique is recently reported which offers, reconfigurability at the filter and architecture levels, in addition to the inherent low complexity offered by the FRM technique. In this paper, we propose a modified architecture to reduce the overall complexity by realizing the prototype filter in the FRM technique by another FRM filter. The hardware implementation of the filter calls for the representation of the filter coefficients in the signed power of two (SPT) space. It is well known that if canonic signed digit (CSD) representation is employed in the SPT space, the hardware complexity can further be significantly reduced. Hence it is proposed in this paper to extend the CSD representation to the FRM based digital filters. The design of the FRM filter in the discrete space degrades the performance and this calls for the use of efficient non-linear optimization techniques. We use genetic algorithm (GA) based optimization which brings forth a near optimal solution. This results in very low power and low complexity FRM based multiplier-less reconfigurable non-uniform channel filters.

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I. Introduction

For the Software Defined Radio (SDR) [Mitola J., 2000], we require a technology which will cater to multi-band, multi-standard and multi-service. This requires systems which are reconfigurable and reprogrammable by software. From the modulated input signal, depending on the communication standards, sub-bands with non-uniform bandwidths are to be extracted by the channel filters in the channelizer in a SDR receiver [Hentschel T et.al, 1999]. The channelizer operates at the highest sampling rate and hence requires low power and low complexity filters. Reconfigurability is an important requirement in an SDR handset. To facilitate this, a reconfigurable structure based on FRM [Lim Y.C., 1986] had been proposed by Smitha K.G. et. al. (2011, 2008) and Mahesh R. et. al (2011, 2007), which offers complexity reduction over the conventional per channel (PC) approach [Hentschel T, 2002]. In the PC approach, a separate filter is used for each communication standard. The low complexity reconfigurable architecture proposed by [Smitha K.G. et. al. (2011, 2008) and Mahesh R. et. al (2011, 2007) is to realize channels with non-uniform bandwidths and is based on the frequency response masking (FRM) method. The same prototype filter but separate interpolation factors and masking filters are used to derive different non-uniform bandwidth channels from the wideband input. So this is a very useful method to get non-uniform sharp transition-width channels and can be extended to realize sharp non-uniform filter banks with low complexity. The complexity of these reconfigurable filter banks can still be reduced by employing multistage masking instead of single stage masking whenever the order of the masking filters is high as proposed by Smitha K.G. et. al. (2011) and Mahesh R. et. al (2007). The problem with this method is that a high order prototype filter also increases the overall complexity. To address this problem, we use a modified architecture in which, if the order of the prototype filter estimated using Bellanger's equation [Bellanger M., 1981] is very high, we can realize the prototype filter also as an FRM filter [Lim Y.C., 1986]. Design example shows that this method offers 98.77% complexity reduction when compared to the PC approach and 53.01% complexity reduction when compared to the approach proposed by Smitha K.G. et. al. (2011) and Mahesh R. et. al (2007). By suitably combining appropriate low pass channels, we can get sharp transition-width band-pass channels also.

The filters designed using Parks-McClellan approach [Parks T. et.al, 1972, James H. et.al, 2005] result in filter coefficients with infinite precision. But for hardware implementation, they have to be represented using finite number of bits. The signed power-of-two (SPT) system [Yong Ching Lim D.L. et.al, 1999] is a representation in which the multiplier coefficients are represented using few non-zero bits. Thus the multiplication operation can be done using a series of shift and add operations. FIR filters with SPT representation is named as multiplier-less digital filters. Canonic Signed Digit (CSD) representation [Reid M. Hewitt et. al, 2000] is a special case of the SPT system. If we restrict the number of non-zero bits in the representation of the filter coefficients, the number of non-zero partial products needed to realize the filter will be reduced and hence the switching activity. Since the power consumption is directly proportional to the switching activity, power consumption also reduces.

But there will be degradation in the magnitude response of the filter due to the restricted number of SPT terms in the representation of the filter coefficients. Degradation in the frequency response due to the truncated FIR filter coefficients can be improved using suitable optimization techniques resulting in low power and low complexity FRM based multiplier-less reconfigurable non-uniform channels. Genetic algorithm for optimizing the CSD represented FRM filter is reported in [Patrick Mercier et.al, 2007], [Yu Y J and Y.C.Lim, 2002] and [Kilambi S. and B. Nowrouzian, 2006]. In this paper, we use ternary coded genetic algorithm [Manoj V.J. and E. Elias, 2009, 2009] for improving the performance of the multiple channel CSD represented filter architecture.

The paper is organized as follows. Section II gives an overview of the single stage and multi stage FRM method. Section III gives an insight into the reconfigurable architecture based on FRM. Section IV explains the CSD technique. Section V gives an overview about genetic algorithm based optimization. A design example and MATLAB simulation results are presented in section VI. Section VII compares the number of multipliers in the proposed method with the existing methods. Section VIII illustrates the method of obtaining band pass channels from low pass channels. Section IX concludes the paper.

II. Overview of Frm Approach

a) Single Stage FRM

Let $H(z)$ be the transfer function of the desired FIR low pass filter with pass band and stop band edge frequencies f_p and f_s respectively. In the FRM technique, the overall sharp transition width filter is composed of many sub-filters of wide transition width. If $H_a(z)$ represents the transfer function of a low pass linear phase filter, its complementary filter $H_c(z)$ can be expressed as given below

$$H_c(z) = z^{-(N-1)/2} - H_a(z) \tag{1}$$

If $H_a(z)$ and $H_c(z)$ are interpolated with a factor M, $H_a(z^M)$ and $H_c(z^M)$ are obtained, whose transition-width is 1/M times the transition-width of $H_a(z)$ i.e. $(f_{as}-f_{ap})/M$. The filters $H_a(z^M)$ and $H_c(z^M)$ are cascaded to the masking filters $H_{Ma}(z)$ and $H_{Mc}(z)$ respectively, which suppress the unwanted images of $H_a(z^M)$ and $H_c(z^M)$. Thus, the transfer function of the overall FIR filter $H(z)$ [Lim Y.C., 1986] is given by

$$H(z) = H_a(z^M)\, H_{Ma}(z) + H_c(z^M)\, H_{Mc}(z) \tag{2}$$

The structure of the FRM FIR filter is given in Fig. 1 [Lim Y.C., 1986].

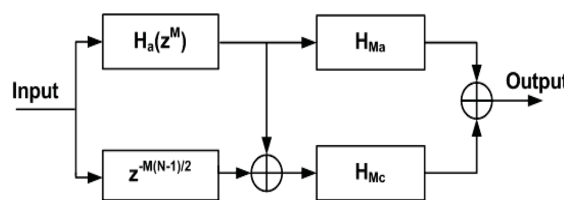


Fig.1 : Basic FRM Filter Architecture

The transition width of the overall filter H(z) is 1/M times the transition-width of $H_a(z)$ i.e. $(f_{as} - f_{ap})/M$. The design steps for the sub-filters are given below

$$m = \lfloor f_p M \rfloor \quad f_{ap} = f_p M - m \quad f_{as} = f_s M - m \tag{3}$$

$$f_{map} = f_p \qquad f_{mas} = (m+1-f_{as})/M$$
$$f_{mcp} = (m-f_{ap})/M \qquad f_{mcs} = f_s \tag{4}$$

Where $\lfloor x \rfloor$ denotes the largest integer less than x, M is the interpolating factor, f_p and f_s respectively are the pass band and stop band frequencies of the final filter $H(z)$. f_{ap} and f_{as} are the pass band and stop band frequencies respectively of the prototype filter $H_a(z)$. f_{map} and f_{mcp} are the pass band frequencies and f_{mas} and f_{mcs} are the stop band frequencies of the masking filters $H_{Ma}(z)$ and $H_{Mc}(z)$ respectively. The frequency responses of each filter are given in Fig. 2.

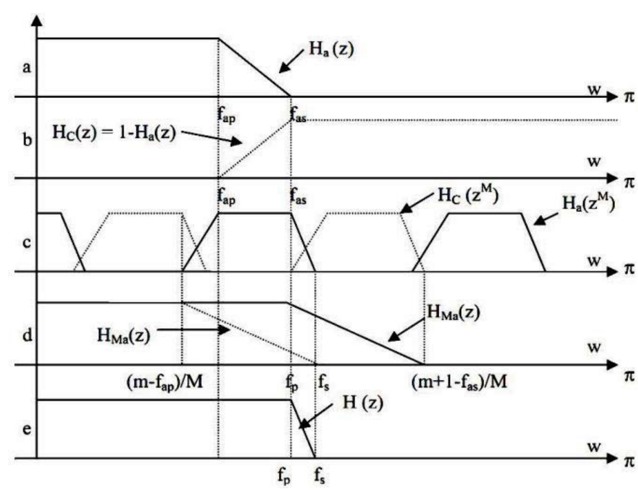


Fig.2 : Frequency response illustration of FRM approach

b) Multistage FRM

If we require a prototype filter with sharp transition width, its order becomes very high and this increases the overall filter complexity. The length of any filter can be estimated using Bellanger's equation [M. Bellanger, 1981] as given below.

$$N = (-2\log(10\delta_1\delta_2)/3\Delta f) - 1 \tag{5}$$

Where δ_1 and δ_2 are the peak pass-band and stop-band ripple magnitudes respectively, and Δf is the normalized transition-bandwidth. If the length of the prototype filter estimated using the above equation is high, multi-stage FRM filter implementation can be used [Lim Y.C., 1986], [Lim Y. C and Lian Y, 1993], [Tapio Saramaki and Yong Ching Lim, 2003] [Yli- Kaakinen J. et.al., 2011, 2004] instead of single stage FRM filter. Thus the complexity of the overall filter can be reduced. The structure of a two stage FRM filter is given in Fig. 3.

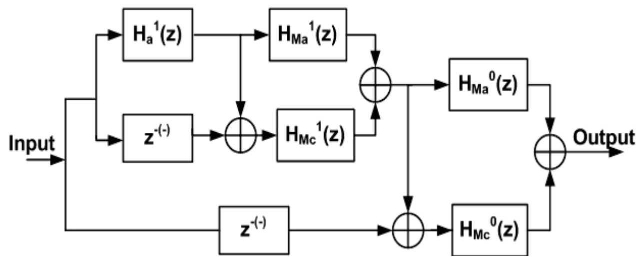


Fig.3 : Structure of two stage FRM filter

III. Reconfigurable Filters Based on Frm

a) Reconfigurable filters with single stage masking

The FRM approach can be extended to more number of channels to form the reconfigurable non-uniform bandwidth channels [Smitha K.G. et. al., 2011, 2008 and Mahesh R. et. al, 2011, 2007]. In this technique, the reconfigurability is achieved using the same prototype filter for all the channels. Different interpolation factors and masking filters are used to derive the different channels. For example, suppose we need two channels with pass band frequencies f_{p1} and f_{p2} and stop band frequencies f_{s1} and f_{s2} respectively. Let f_{ap} and f_{as} be the pass band and stop band frequencies respectively of the prototype filter which is the same for both channels. The prototype filter with specifications f_{ap} and f_{as} and interpolation factors M_1 and M_2 for channels 1 and 2 respectively can be designed by iterating the equation given below for different values of M_1 and M_2 [Smitha K.G. et. al., 2011, 2008 and Mahesh R. et. al, 2011, 2007].

$$f_{ap} = f_{p1}M_1 - \lfloor f_{p1}M_1 \rfloor = f_{p2}M_2 - \lfloor f_{p2}M_2 \rfloor$$

$$f_{as} = f_{s1}M_1 - \lfloor f_{s1}M_1 \rfloor = f_{s2}M_2 - \lfloor f_{s2}M_2 \rfloor \qquad (6)$$

The masking filters for each channel can be designed using the equations given in (4). The structure of the two channel reconfigurable filters is given in Fig 4.

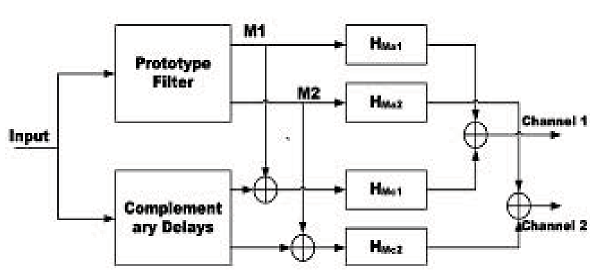


Fig. 4 : Two-channel single-stage architecture

H_{Ma1} and H_{Mc1} are the masking filters for channel 1 and H_{Ma2} and H_{Mc2} are the masking filters for channel 2. M_1 and M_2 are the interpolation factors for channel 1 and channel 2 respectively.

b) Reconfigurable filters with multi-stage masking

If the interpolation factor is high, the complexity of the masking filters also becomes high. This can be reduced using multi-stage masking instead of single stage masking. In multi-stage masking, the interpolation factor is factorized and masking is implemented in multiple stages where, in each stage masking filters are implemented for lower interpolation values which are factors of the overall interpolation factor [Smitha K.G. et. al., 2011 and Mahesh R. et. al, 2007]. The architecture of the reconfigurable filters using two stage masking is shown in Fig 5. Here, the prototype filter response is interpolated by factors M_1 and M_2 respectively. H_{Ma1} and H_{Mc1} are the masking filters designed for interpolation factor M_1. If M_2 can be factorized as $M_{21}*M_{22}$, then masking can be implemented in two stages where M_{21} and M_{22} are the interpolation factors of the first and second stage respectively. H_{Ma21} and H_{Mc21} are the first stage masking filters designed using equations (4) for interpolation factor M_{21} and they are interpolated with value M_{22}. H_{Ma22} and H_{Mc22} are the second stage masking filters designed for interpolation factor M_{22}.

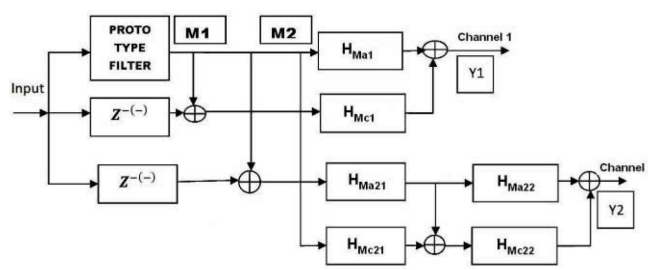


Fig. 5 : Architecture of two stage masking

c) Proposed Method - Reconfigurable filters with multistage FRM and multistage masking

In the architecture given in section IIIB, the problem with higher order prototype filter is not addressed. If the order of the prototype filter is high, the overall complexity of the multiple channel filter structure also will be high. In the proposed method, we modify the architecture in section IIIB by implementing the prototype filter using FRM as discussed in section IIB. This is shown in Fig. 6. Using this method, sharp transition-width channels with very low complexity can be obtained. Also since the number of filters is increased, we get improved pass band and stop band characteristics. By selecting appropriate low pass channels for subtraction, sharp transition-width band-pass channels with very low complexity can also be obtained. This is illustrated in section VIII.

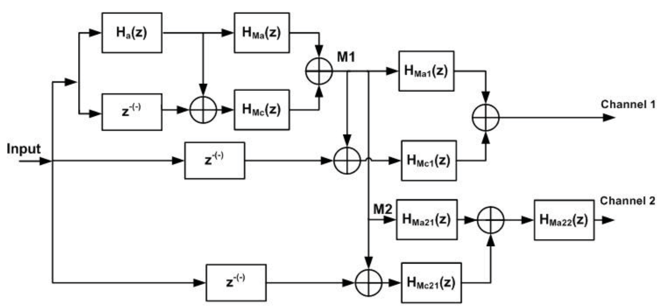


Fig. 6 : Reconfigurable non-uniform bandwidth filters with multistage masking and multistage FRM

IV. Canonic Signed Digit Representation

Any FIR filter can be represented [Zhangwen Tang et.al, 2002] as

$$y(n) \;=\; \sum_{k=0}^{N-1} h(k)x(n-k) \qquad (7)$$

Where N is the length of the FIR filter, h(k) are the filter coefficients and x(n) is the input signal. The FIR filter implementation consists of multiplications, which are realized by shifters and adders. For the multiplication of the two N-bit numbers represented in the 2's complement form, in the worst case, N shifters and N-1 adders are needed. The number of non-zero partial product additions is determined by the number of non-zero bits in the filter coefficient. As the number of non-zero bits is reduced, the partial product additions are also reduced. CSD representation is a unique representation of the filter coefficients with minimum number of non-zero bits [Reid M. Hewitt et. al, 2000 and Zhangwen Tang et. al, 2002]. A fractional number q is represented in CSD format as

$$q \;=\; \sum_{i=1}^{W} c_i 2^{R-i} \qquad (8)$$

Where $c_i = \{-1, 1, 0\}$ and W is the word length of the CSD number. Since, this encoding uses -1, 0 and 1 digits, it is called ternary coding. No adjacent digits in the CSD representation can be non-zero i.e. $c_i * c_{i-1} = 0$, where c_i is the i^{th} digit in the CSD representation. The maximum number of non-zero digits in the CSD representation of an n-bit number is floor(n/2), compared with n bits in the 2's complement representation. The number of adders/subtractors needed to realize CSD represented filter coefficient will be floor(n/2)-1.

V. Genetic Algorithm

Several optimization methods are proposed for the optimization of infinite precision FRM filter in which either separate optimization of sub-filters is done or joint optimization of the filters is done. The sub-filters designed using linear programming in the paper [Lim Y. C., 1986] reduced the error in the pass band and stop band of the overall FRM filter. Another approach to reduce the error in the pass band and stop band of the overall FRM filter is to design the sub-filters using Remez algorithm as proposed by Tapio Saramaki and Yong Ching Lim, 2003.

The optimization of the FRM filter in the discrete space is a complicated process and so efficient non-linear optimization techniques need to be used. The classical gradient based optimization techniques cannot be directly applied to this problem because, here the search space consists of integers. In this context, meta-heuristic algorithm is a good optimization tool as the proper selection of the parameters with respect to a particular design problem can bring forth global solution.

Genetic algorithms (GA) have been established as a good alternative for the optimization of multimodal, multi dimensional problems. This is a population based evolutionary algorithm where, in each iteration, candidate solutions are generated using genetic operations like reproduction, crossover and mutation. The use of GA for the optimization of the frequency responses of the CSD represented filters is employed by Yu Y.J. and .C. Lim, 2002, Samadi P. and M. Ahmad, 2007, [Fuller A. et.al, 1999 and Uppalapati H. et.al., 2005. Genetic algorithm for the optimization of FRM filter in the SPT space is used by Patrick Mercier et.al, 2007, Yu Y J and Y.C.Lim, 2002 and Kilambi S. and B. Nowrouzian, 2006.

a) Encoding of the optimization variables

Once the infinite precision filters are designed, the coefficients of these filters have to be converted to the CSD representation. As we have discussed in section IV, one way to encode filters in the CSD space is to represent them using ternary coding as proposed by Manoj V.J. and E. Elias, 2009. A look up table is created which has four fields. The four fields are index, CSD numbers, decimal equivalents and the number of SPT terms. In our work, the maximum allowed precision is 12 bits.

When crossover and mutation operations are performed on these ternary coded coefficients, the canonical property of the CSD representation of the coefficients may be lost. To ensure the canonical property of the filter coefficient representation, many restoration algorithms are proposed [Fuller A. et.al, 1999, Uppalapati H. et.al., 2005]. But these restoration algorithms increase the computational complexity of GA. A ternary coding based GA and a simple modified restoration algorithm is proposed by Manoj V.J. and E. Elias, 2009. We use this GA and restoration algorithm in our work.

b) Objective function for the design of the CSD coefficients of filters

When the filter coefficients are rounded to the nearest CSD number with restricted number of SPT terms, the stop band and pass band properties get degraded. So the filter coefficients in the CSD format have to be optimized to improve the pass band and stop band characteristics. In this paper, we use genetic algorithm based optimization technique to improve these characteristics. In our design, the CSD filter coefficients can use any number of SPT terms. But the total number of SPT terms used is restricted. This method gives more flexibility over restricting each coefficient with fixed number of SPT terms [Manoj V.J. and E. Elias, 2009].

The pass band and stop band characteristics are taken care of by minimizing the pass band ripple and maximizing the stop band attenuation. So, the objective function used in our work is given by

Minimize

$$\text{Max}\ \{\,|(|H(e^{j\omega})|-1)|,\ \omega<\omega_p \quad |H(e^{j\omega})|,\ \ \omega>\omega_s\} \tag{9}$$

ω_p and ω_s are the pass band and stop band cut off frequencies respectively. So the objective function is to optimize the above values under the constraint that the total number of SPT terms used is restricted.

c) Optimization of multiple channel filter architecture using ternary coded GA

Since all the filters in our architecture have linear phase property, only the first half of the CSD represented filter coefficients are optimized and the other half of the filter coefficients are obtained using linear phase property. The genetic algorithms for optimizing the multiple channel filter architecture have two phases. Since we have multiple channels in our architecture, each channel has to be optimized separately. Also, since all the channels are derived from the same prototype filter outputs, we have to separately optimize the masking filters of each channel and the prototype filter. So, in the first phase, the prototype filter is optimized and in the second phase the masking filters of each channel are optimized.

Phase 1: Optimization of prototype filter: The basic steps in GA to optimize the filters are given in Fig 7 and are explained below [Randy L.H. and Sue E.H, 2004].

a) Initialization: The initial chromosome is generated by concatenating the first half of the continuous filter coefficients of the prototype filter. If the prototype filter is implemented by another FRM filter, the initial chromosome is generated by concatenating the first half of the continuous filter coefficients of all the sub-filters constituting the FRM prototype filter.

This initial chromosome has been rounded to the nearest CSD representation with maximum number of non-zero bits, which is chosen as 6 in this work. By changing the initial chromosome by random perturbations, a population pool of N-1 chromosomes is generated, where N is the population size. The coefficients of all these N-1 filters are then converted to the CSD representation with restricted number of non-zero bits. These N-1 perturbed chromosomes along with the initial non-perturbed chromosome will make the initial population of size N.

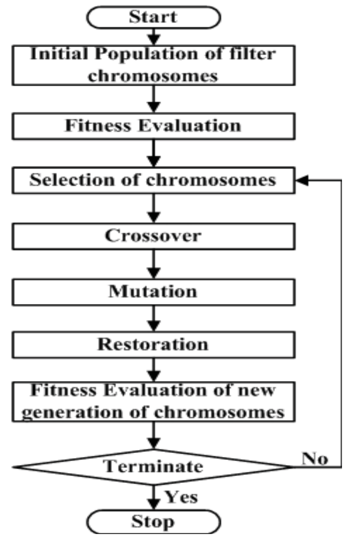


Fig. 7 : GA design flow for filter optimization

b) Fitness Evaluation of initial population : Each chromosome in the population is evaluated using the objective function given in (9) and they are ranked.

c) Selection : The best chromosomes based on ranking are selected from the population and they form the mating pool. In this paper, Roulette wheel rank selection [Randy L.H. and Sue E.H, 2004] is used for selecting the mates for cross over and reproduction.

d) Crossover : Offspring for the next generation are generated by exchanging the genes of parents. We use two-point crossover in our work, in which the genes are swapped between the parents between the two selected points.

e) Mutation : Some of the best solutions from the current population are propagated to the next generation without any change and the remaining chromosomes are mutated with some new information and propagated to the next generation. Due to cross over and mutation, the canonic property of the filter coefficients may be violated. The canonic property is retained using a simple restoration algorithm in which when two consecutive non-zero terms come, one of them is made zero [Manoj V.J. and E. Elias 2009].

f) Fitness Evaluation of New Population: Each chromosome in the new population is evaluated using the objective function given in (9) and they are ranked.

The steps from c to f are repeated until a maximum number of iterations are reached and then GA is terminated. When it is terminated, the best chromosome is taken from the population and is decoded to get the optimum CSD represented filter. The above steps have to be repeated for the prototype filter and masking filters separately.

Phase 2: Optimization of masking filters of each channel: After the prototype filter is optimized, the masking filters of each channel need to be optimized separately using the same objective function given in (9). In this phase, we adopt the joint optimization of the masking filters in which the initial chromosome is generated by concatenating the first half of the continuous filter coefficients of all the masking filters of a channel. The steps of GA algorithm are repeated until the maximum numbers of iterations are reached. This same optimization process has to be repeated for the masking filters of all the channels.

VI. Design Example and Simulation Results

The design example shown here compares the proposed method discussed in section IIIC with the methods discussed in section IIIA and IIIB. In this example, two channels of non-uniform bandwidth are designed for CDMA (Code Division Multiple Access) and PHS (Personal Handy Phone System) standards. The specifications for each channel are given below.

First Channel (CDMA)

Pass band (PB) frequency: 1250 kHz

Maximum pass band ripple: 0.1 dB

Stop band (SB) frequency: 1251 kHz

Minimum stop band attenuation: 40 dB

Sampling Rate: 6 MHz

Normalized PB frequency, $f_{p1} = 1250/6000 = 0.2083$

Normalized SB frequency, $f_{s1} = 1251/6000 = 0.2085$

Second Channel (PHS)

Pass band frequency: 300 kHz

Maximum pass band ripple: 0.1 dB

Stop band frequency: 301 kHz

Minimum stop band attenuation: 40 dB

Sampling Rate: 6 MHz

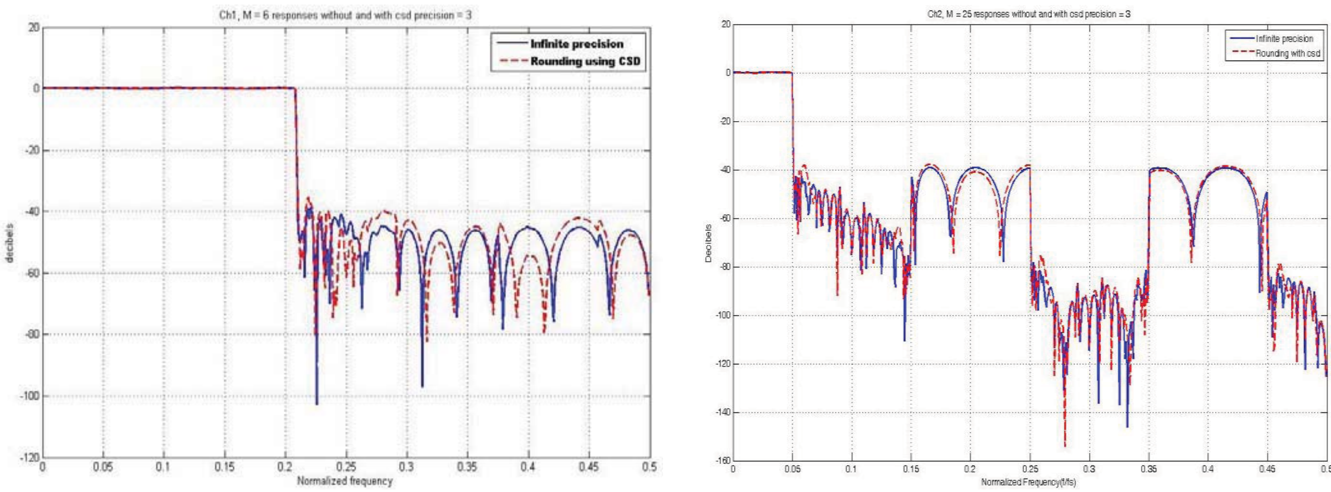
Normalized PB frequency, $f_{p2} = 300/6000 = 0.05$

Normalized SB frequency, $f_{s2} = 301/6000 = 0.0502$

The architecture for this non-uniform bandwidth two channel implementation is shown in Fig. 6. Using equation (6) and iterating for different values of M_1 and M_2, we obtain $M_1 = 6$, $M_2 = 25$, $f_{ap} = 0.25$ and $f_{as} = 0.255$. When these values are substituted in Bellanger's equation given in (5), the length of the prototype filter is obtained as 391 and the number of multipliers needed is 196. Since the length of the prototype filter is high, it is implemented as FRM filter and the number of multipliers needed for FRM implementation is then found to be 60. The CDMA filter response is implemented using single stage masking since its interpolation factor is small ($M_1 = 6$). For PHS filter response, the interpolation factor is 25 and can be factorized into 5 and 5. So, PHS filter is implemented using two stages masking with interpolation factor 5 in each stage. The length of the masking and masking complementary filters H_{Ma1} and H_{Mc1} of channel 1 is 24 and 23 respectively. The length of the first stage masking and masking complementary filters H_{Ma21} and H_{Mc21} of channel 2 is 20 and 19 respectively. The length of the second stage masking filter H_{Ma22} of channel 2 is 20. The total number of multipliers needed for the proposed method is 117 compared to the method proposed by Smitha K.G. et. al. (2011) and Mahesh R. et. al (2007) where 249 multipliers are needed. Thus, there is a 53.01% reduction in the number of multipliers. The peak pass band ripple and minimum stop band attenuation obtained for channel 1 and channel 2 when the prototype filter was implemented directly using Parks - McClellan method as in Fig. 5 and when it was implemented using FRM method as in Fig. 6, are shown in Table 1. From this table, it is seen that when the prototype filter was implemented using FRM method, the stop band characteristics are also improved compared to the direct implementation using Parks-McClellan method.

Table 1 : Performance comparison

Methods	Channel 1		Channel 2	
	Max. Pass Band Ripple (dB)	Min. Stop Band Attenuation (dB)	Max. Pass Band Ripple (dB)	Min. Stop Band Attenuation (dB)
Reconfigurable two channel filters with two stage masking and direct implementation of prototype filter	0.213	37.28	0.2268	37.98
Reconfigurable two channel filters with two stage masking and FRM prototype filter	0.178	38.66	0.2306	39.14



(a) Channel 1 (b) Channel 2

Fig. 8 : Frequency response of channel 1 ($M_1 = 6$) and channel 2 ($M_2 = 25$) using infinite precision filters and CSD represented filters with maximum 3 SPT terms

Next, the filters are represented using CSD with restricted number of bits. The restriction in the number of non-zero bits in the CSD representation is varied from 2 to 6 and the performance is compared. The frequency responses of the infinite precision filters and CSD represented filters with maximum three SPT terms, for channel 1 and channel 2 are shown in Fig 8a and 8b respectively.

In Table 2, the peak pass band ripple and minimum stop band attenuation of the 2 channels for the infinite precision filters and CSD represented filters using different number of non-zero bits are shown. From Table 2, it can be seen that when the number of non-zero bits for the CSD representation is restricted to 3, for the first channel, the peak pass band ripple is increased by 0.0215 dB and minimum stop band attenuation is degraded by 3.22 dB. For the second channel, the stop band attenuation is degraded by 1.33 dB compared to the infinite precision filter response. So we have to optimize the CSD represented filter responses to get better responses for channel 1 and 2. Here, we have employed GA optimization to improve the pass band and stop band responses.

For the multiple channel filter architecture, optimization includes two phases. In the first phase, the coefficients of the prototype filter are optimized which is implemented as an FRM filter. The FRM prototype filter is made up of sub-filters and each of these continuous filter coefficients has linear phase characteristics. So for optimizing the coefficients of a prototype filter, the genes corresponding to the first 30 genes of the FRM sub-filter, the first 16 genes of the masking filter and the first 16 genes of the masking complementary filter are concatenated, to generate the chromosomes in the initial population. The maximum allowed number of SPT terms is taken as 186 so that the average number of SPT terms in a filter coefficient is 3. The different parameters used for the optimization of the prototype filter are given below:

Number of Iterations = 500

Population Size = 50

Table 2 : Performance comparison when max. no: of SPT terms are varied

	Channel 1		Channel 2	
Coefficients	**Max. Pass Band Ripple(dB)**	**Min. Stop Band Attenuation (dB)**	**Max. Pass Band Ripple(dB)**	**Min. Stop Band Attenuation (dB)**
With Infinite precision	0.178	38.66	0.2306	39.14
Number of non-zero bits = 2	0.2965	32.07	-0.4419	28.24
Number of non-zero bits = 3	0.1995	35.44	-0.2144	37.81
Number of non-zero bits = 4	0.1944	39.2	0.2254	38.44
Number of non-zero bits = 5,6	0.1943	39.58	0.2171	38.76

Number of population members that survive each generation = 1

Mutation Rate = 0.02

Number of the best population which is kept without change during mutation (Elite Count) = 10

In the second phase, the coefficients of the masking filters of each channel are optimized. Each of these continuous filter coefficients has linear phase characteristics. Therefore, for optimizing the coefficients of the masking filters of channel 1, the genes corresponding to the first 12 genes of the masking filter and the first 12 genes of the masking complementary filter are concatenated to generate the chromosomes in the initial population. The maximum allowed number of SPT terms is taken as 72 so that the average number of SPT terms in a filter coefficient is 3. Similarly for optimizing the coefficients of the masking filters of channel 2, the genes corresponding to the first 10 genes of the first stage masking filter, the first 10 genes of the first stage masking complementary filter and the first 10 genes of the second stage masking filter are concatenated to generate the chromosomes in the initial population. The maximum allowed number of SPT terms are taken as 90 so that the average number of SPT terms in a filter coefficient is 3. The different parameters used for the optimization of the masking filters of both channels are given below:

Number of Iterations = 500

Population Size = 50

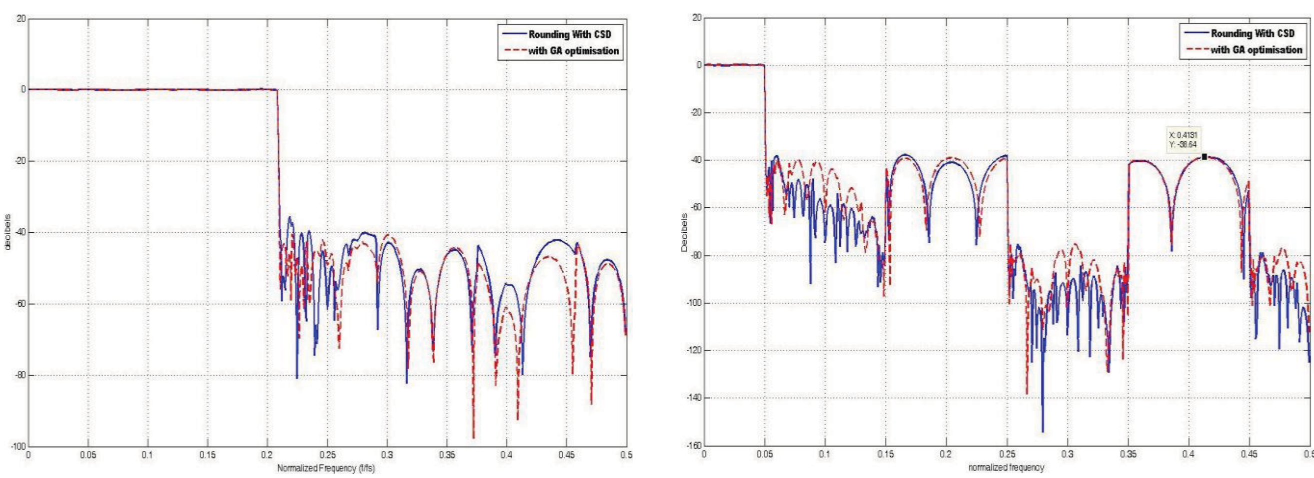
Number of population members that survive each generation = 5

Mutation Rate = 0.2

Number of the best population which is kept without change during mutation (Elite Count) = 10

The frequency responses of channel 1 and channel 2 after GA optimization are given in Fig 9a and 9b respectively.

The peak pass band ripple and minimum stop band attenuation obtained for channel 1 and channel 2 outputs after GA optimization of the prototype filter and the masking filters with maximum 3 SPT terms are given in Table 3. From the table it is clear that, when we employed optimization, the pass band responses of channel 1 and channel 2 are improved by 0.0233 dB and 0.0723 dB respectively and the stop band responses of channel 1 and channel 2 are improved by 4.92 dB and 0.83 dB respectively.



(a) Channel 1 (b) Channel 2

Fig. 9 : Frequency response of channel 1 (M_1 = 6) and channel 2 (M_2 = 25)with CSD represented filters with maximum 3 SPT terms before and after GA optimization

Table 3 : Performance comparison of CSD represented filters with maximum 3 SPT terms before and after GA optimization

.	Channel 1		Channel 2	
Coefficients	Max. Pass Band Ripple(dB)	Min. Stop Band Attenuation (dB)	Peak Pass Band Ripple(dB)	Min. Stop Band Attenuation (dB)
With Infinite precision	0.178	38.66	0.2306	39.14
With CSD representation	0.1995	35.44	-0.2144	37.81
With CSD representation and GA	0.1762	40.36	-0.1421	38.64

If the number of non-zero bits is reduced to 2, further reduction in complexity is obtained. But from Table 2, it can be seen that when the number of non-zero bits for the CSD representation is restricted to 2, the peak pass band ripple is increased by 0.1185 dB and minimum stop band attenuation is degraded by 6.59 dB for the first channel. For the second channel, the peak pass band ripple is increased by 0.2113 dB and the minimum stop band attenuation is degraded by 10.9 dB compared with the infinite precision filter responses. The different parameters used for the optimization of the prototype filter are given below:

Number of Iterations = 500

Population Size = 50

Number of population members that survive each generation = 1

Mutation Rate = 0.02

Number of the best population which is kept without change during mutation (Elite Count) = 5

The different parameters used for the optimization of the masking filters of channel 1 and channel 2 are given below:

Number of Iterations = 500

Population Size = 50

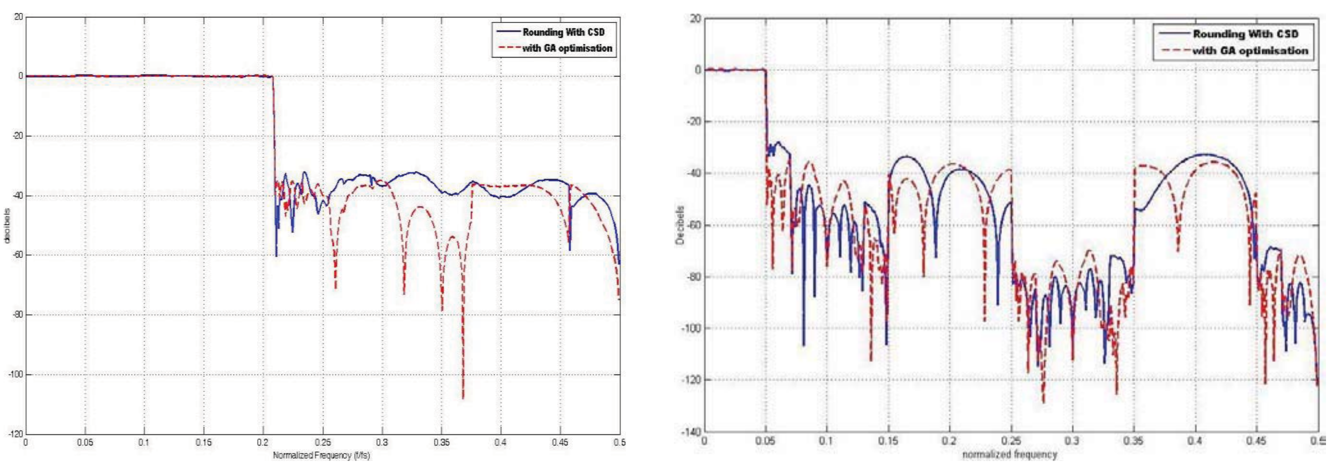
Number of population members that survive each generation = 15

Mutation Rate = 0.1

Number of the best population which is kept without change during mutation (Elite Count) = 20

The frequency responses of channel 1 and channel 2 after GA optimization are given in Fig 10a and 10b respectively.

The peak pass band ripple and minimum stop band attenuation achieved for channel 1 and channel 2 outputs after GA optimization of the prototype filter and masking filters are given in Table 4. From the table it is clear that, the pass band responses of channel 1 and channel 2 are improved by 0.0832 dB and 0.2806 dB respectively and stop band response of channel 1 and channel 2 are improved by 2.6 dB and 6.34 dB respectively, after optimization.



(a) Channel 1 (b) Channel 2

Fig. 10 : Frequency response of channel 1 ($M_1 = 6$) and channel 2($M_2 = 25$)with CSD represented filters with maximum 2 SPT terms before and after GA optimization

Table 4 : Performance comparison of CSD represented filters with maximum 2 SPT terms before and after GA optimization

	Channel 1		Channel 2	
Coefficients	Peak Pass Band Ripple(dB)	Minimum Stop Band Attenuation (dB)	Peak Pass Band Ripple(dB)	Minimum Stop Band Attenuation (dB)
With Infinite precision	0.178	38.66	0.2306	39.14
With CSD representation	0.2965	32.07	-0.4419	28.24
With CSD representation and GA optimization	0.2133	34.67	0.1613	34.58

VII. Complexity Comparision

Here, we compare the number of multipliers used to design channel 1 and channel 2 filters. The number of multipliers used to design any filter is given by

$$f(N) = (N + 1)/2 \text{ if N is odd}$$
$$(N/2) + 1 \text{ If N is even}$$

Where f(N) denotes the total number of multipliers needed for the implementation of an FIR filter with order N. If N_a is the order of the prototype filter $H_a(z)$, N_{Ma} is the order of the masking filter $H_{Ma}(z)$, and N_{mc} is the order of the masking complementary filter $H_{Mc}(z)$, the number of multiplications, π, required to implement the overall FIR filter is given by

$$\pi = f(N_a) + f(N_{ma}) + f(N_{mb})$$

In Table 5, the complexity comparison of the proposed method with the existing method in terms of multipliers is shown. We can see that the proposed method offers a 53.01% reduction over the method discussed in section IIIB.

VIII. Band-pass Channels From Low Pass Channels

In this section, the method of extracting band pass channels from the designed low pass channels is shown. To illustrate this, we have modified the structure shown in Fig. 6 to a three channel structure by adding one more channel with the following specifications. The frequency responses of all the three low pass channels are given in Fig 11(a-c).

Third Channel

Pass band frequency: 900 kHz

Maximum pass band ripple: 0.1 dB

Stop band frequency: 901 kHz

Minimum stop band attenuation: 40 dB

Sampling Rate: 6 MHz

Normalized PB frequency, fp2 = 900/6000 = 0.15

Normalized SB frequency, fs2 = 901/6000 = 0.1502

The band-pass responses are shown in Fig 11(d-f). The band-pass response shown in Fig. 11d is derived by combining the first and second channel low pass responses. The band-pass response shown in Fig. 11e is obtained by combining the first and third low pass channel responses. The band-pass response shown in Fig. 11f is obtained by combining the second and third channel responses.

Table 5 : Complexity comparison of various methods with the proposed method for two channel filter architecture

No:	Method	Channels	No. Of Multipliers				% saving in multipliers
			H_a	H_{Ma}	H_{Mc}	Total	
1	PC method	(First channel)		4751		9502	
		(Second channel)		4751			
2	PC method using FRM Filters	(First channel)	135	88	65	573	93.97%
		(Second channel)	143	70	72		
3	Reconfigurable two channel filters with single-stage masking	(First channel)	192	13	12	314	96.7% w.r.t 1 45.2% w.r.t 2
		(Second channel)		49	48		
4	Reconfigurable two channel filters using two stage masking for second channel filter	(First channel)	192	13	12	249	97.38% w.r.t 1 56.54% w.r.t 2 20.7% w.r.t 3
		(Second channel first stage masking)		11	10		
		(Second channel second stage masking)		11			
5	Reconfigurable two channel filters with two stage masking and FRM prototype filter	(First channel)	60	13	12	117	98.77% w.r.t 1 79.58% w.r.t 2 62.74% w.r.t 3 53.01% w.r.t 4
		(Second channel first stage masking)		11	10		
		(Second channel second stage masking)		11			

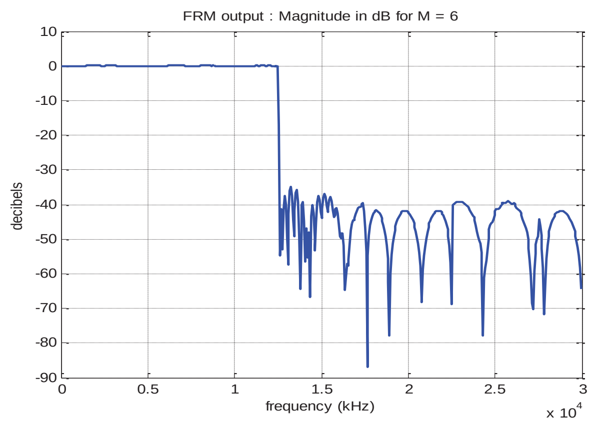


Fig.11a : First low pass channel response

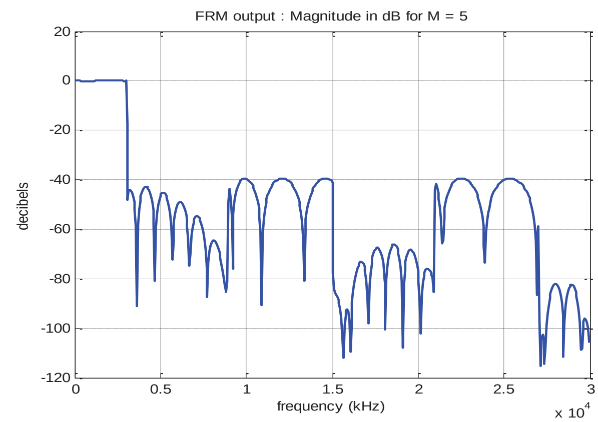


Fig.11b : Second low pass channel response

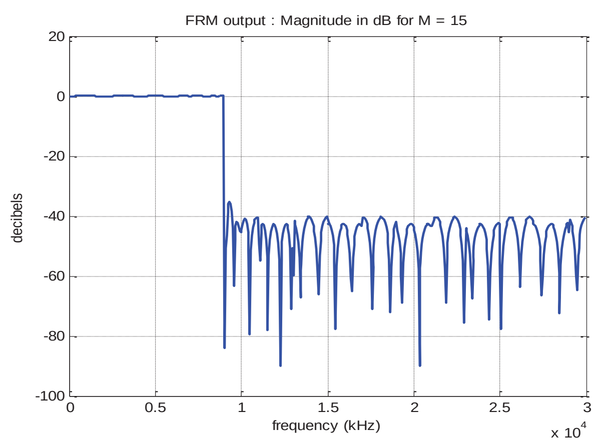


Fig.11c : Third low pass channel response

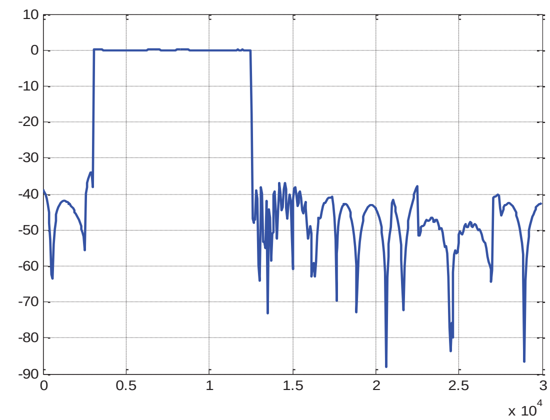


Fig.11d : First Band-pass response

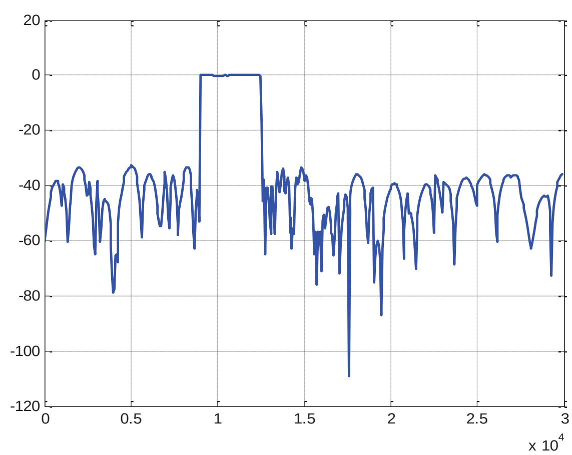


Fig.11e : Second Band-pass response

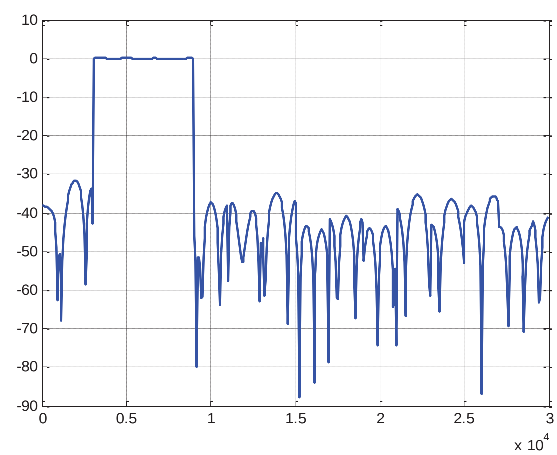


Fig.11f : Third Band-pass response

IX. Conclusion

The existing architecture with low power, low complexity and reconfigurability for software defined radio is modified in this paper. The prototype filter in the FRM structure is replaced by another FRM structure. The filter coefficients are represented in the signed power of two spaces, where CSD representation is employed. Thus we get multiple channel architecture with very low complexity and low power which is ready for hardware implementation. Since the design of the filter in the discrete space degrades the performance, the response is optimized by a modified genetic algorithm which results in near optimal solutions. This leads to the implementation of low power, low complexity multiplier-less, reconfigurable, non-uniform channel filters. Since different channels correspond to different communication standards, different objective functions and different optimization techniques may be used which may lead to better performance. This architecture can be extended to more number of channels also.

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Mathematical Analysis of Pulse Tube Cryocoolers Technology

By Shashank Kumar Kushwaha , Amit medhavi & Ravi Prakash Vishvakarma

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Abstract - The cryocoolers are being developed for use in space and in terrestrial applications where combinations of long lifetime, high efficiency, compactness, low mass, low vibration, flexible interfacing, load variability, and reliability are essential. Pulse tube cryocoolers are now being used or considered for use in cooling infrared detectors for many space applications. In the development of these systems, as presented in this paper, first the system is analyzed theoretically. Based on the conservation of mass, the equation of motion, the conservation of energy, and the equation of state of a real gas a general model of the pulse tube refrigerator is made. The use of the harmonic approximation simplifies the differential equations of the model, as the time dependency can be solved explicitly and separately from the other dependencies. The model applies only to systems in the steady state. Time dependent effects, such as the cool down, are not described. From the relations the system performance is analyzed. And also we are describing pulse tube refrigeration mathematical models. There are three mathematical order models: first is analyzed enthalpy flow model and heat pumping flow model, second is analyzed adiabatic and isothermal model and third is flow chart of the computer program for numerical simulation. These mathematical reviews describe cryocoolers working and operation.

Keywords : *cryocooler, various applications, different types of cryocooler and mathematical analysis.*

GJRE-F Classification : *FOR Code: 010299*

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Strictly as per the compliance and regulations of:

Mathematical Analysis of Pulse Tube Cryocoolers Technology

Shashank Kumar Kushwaha [α], Amit medhavi [σ] & Ravi Prakash Vishvakarma [ρ]

Abstract - The cryocoolers are being developed for use in space and in terrestrial applications where combinations of long lifetime, high efficiency, compactness, low mass, low vibration, flexible interfacing, load variability, and reliability are essential. Pulse tube cryocoolers are now being used or considered for use in cooling infrared detectors for many space applications. In the development of these systems, as presented in this paper, first the system is analyzed theoretically. Based on the conservation of mass, the equation of motion, the conservation of energy, and the equation of state of a real gas a general model of the pulse tube refrigerator is made. The use of the harmonic approximation simplifies the differential equations of the model, as the time dependency can be solved explicitly and separately from the other dependencies. The model applies only to systems in the steady state. Time dependent effects, such as the cool down, are not described. From the relations the system performance is analyzed. And also we are describing pulse tube refrigeration mathematical models. There are three mathematical order models: first is analyzed enthalpy flow model and heat pumping flow model, second is analyzed adiabatic and isothermal model and third is flow chart of the computer program for numerical simulation. These mathematical reviews describe cryocoolers working and operation.

Keywords : cryocooler, various applications, different types of cryocooler and mathematical analysis.

I. Introduction

Cryogenics comes from the Greek word "kryos", which means very cold or freezing and "genes" means to produce. A Cryocooler is closed cycle cooler of a device which is used to cool inside the environment of anything and increasing need in cryogenic temperature in research and high conductivity during the last decade caused a rapid development of cryocoolers. In a country like India [1],

The cost of liquid Helium and liquid Hydrogen is increasing, cryocoolers can play a very important role [1]. Its Refrigeration powers vary from about (0.15 W to 1.75 w). The ability of the device to cool its interior environment depends largely on the thermodynamic properties of the gas circulating through the system. Cryocooler may be classified into different types of pulse tube which called various name, the important factors are discussed that have brought the pulse tube refrigerator to its current position as one of the most promising cryocoolers for a wide variety of applications[2].

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II. Applications

The main requirement is it's cooled below 120k which is use in various applications, area is very large. Cryocoolers are refrigerating machines, which are able to achieve and to maintain cryogenic temperatures [3].

a. Military
1. Infrared sensors for missile guidance & night vision
2. Infrared sensors for surveillance (satellite based)
3. Gamma ray sensors for monitoring nuclear activity

b. Commercial
1. Cryopumps for semiconductor fabrication
2. Superconductors for cellular-phone base stations
3. Superconductors for high-speed communications

c . Medical
1. cooling superconducting magnets for MRI Claude
2. SQUID magnetometers for heart and brain studies
3. Liquefaction of oxygen for hospital and home use

d. Transportation
1. LNG for fleet vehicles
2. Superconducting magnets in maglev trains
3. Infrared sensors for aircraft night vision

e. Energy
1. LNG for peak shaving
2. Superconducting power applications (motors, transformers etc.)
3. Infrared sensors for thermal loss measurements

f . Police and Security
Infrared sensors for night-security and rescue

g. Agriculture and Biology
Storage of biological cells and specimens

III. Classification of Cryocooler

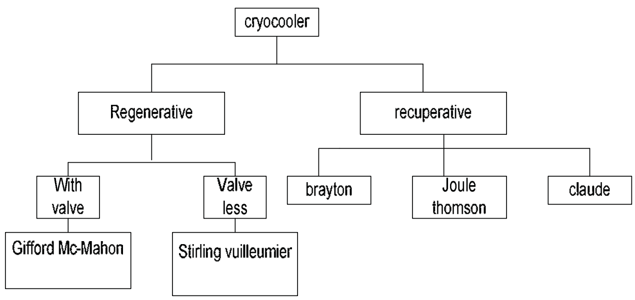

Fig 1 : Schematic classification of various types of cryocooler

a) *Recuperative Cryocoolers*

The recuperative coolers use only recuperative heat exchangers and operate with a steady flow of refrigerant through the system. The compressor operates with a fixed inlet pressure and a fixed outlet pressure. If the compressor is a reciprocating type, it must have inlet and outlet valves (valve compressor) to provide the steady flow. Scroll, screw or centrifugal compressors do not need valves to provide the steady flow [4]. Figure2 shows schematics of the most common recuperative cryocooler cycles. Expansion of the liquid in the JT capillary, orifice, or valve is relatively efficient and provides enough of a temperature drop that little or no heat exchange with the returning cold, expanded gas is required. Thus, a very efficient recuperative heat exchanger is required to reach cryogenic temperatures.

i. *Joule Thomson Cryocoolers*

The Joule-Thomson cryocoolers produce cooling when the high pressure gas expands through a flow impedance (orifice, valve, capillary, porous plug), often referred to as a JT valve. The expansion occurs with no heat input or production of work, thus, the process occurs at a constant enthalpy. The heat input occurs after the expansion and is used to warm up the cold gas or to evaporate any liquid formed in the expansion process [5]. The main advantage of JT cryocoolers is the fact that there are no moving parts at the cold end. The cold end can be miniaturized and provide a very rapid cool down. This rapid cool down (a few seconds to reach 77 K) has made them the cooler of choice for cooling infrared sensors used in missile guidance systems. These coolers utilize a small cylinder pressurized to about 45 M Pa with nitrogen or argon as the source of high pressure gas. Miniature finned tubing is used for the heat exchanger. An explosive valve is used to start the flow of gas from the high pressure bottle. The higher boiling point components must remain a liquid at the lowest temperature [6].

ii. *Brayton Cryocoolers*

In Brayton cryocoolers (sometimes referred to as the reverse Brayton cycle to distinguish it from a heat engine) cooling occurs as the expanding gas does work. Figure shows a reciprocating expansion engine for this purpose, but an expansion turbine supported on gas bearings is more commonly used to give high reliability. According to the First Law of Thermodynamics the heat absorbed with an ideal gas in the Brayton cycle is equal to the work produced.

The Brayton cycle is commonly used in large liquefaction plants. For small Brayton cryocoolers the challenge is fabricating miniature turbo expanders that maintain high expansion efficiency. The expansion engine provides for good efficiency over a wide temperature range, although not as high as some Stirling and pulse tube cryocoolers at temperatures above about 50 K. The low-pressure operation of the miniature Brayton systems requires relatively large and expensive heat exchangers [7].

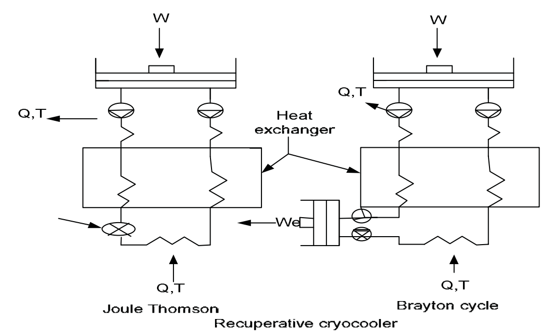

Fig. 2 : Joule Thomson and Brayton cryocooler

b) *Regenerative Cryocoolers*

In regenerative types the refrigerant undergoes an oscillating flow or an Oscillating pressure analogous to an AC electrical system. The compressor and the pressure Oscillator for the regenerative cycles need no inlet or outlet valves. The regenerator has only one flow channel, and the heat is stored for a half cycle in the regenerator matrix, which must have a high heat capacity. The performance of the regenerative type cryocoolers is dependent on the phase difference between the pressure and mass flow rate phases. Helium is the refrigerant of choice for most regenerative type cryocoolers [8].

i. *Stirling Cryocoolers*

In the Stirling cryocooler the second moving component, the displacer, is required to separate the heating and cooling effects by causing motion of the gas in the proper phase relationship with the pressure oscillation. When the displacer is moved downward, the helium gas is displaced to the warm end of the system through the regenerator. The piston in the compressor then compresses the gas, and the heat of compression is removed by heat exchange with the ambient.

Next the displacer is moved up to displace the gas through the regenerator to the cold end of the system. The piston then expands the gas, now located at the cold end, and the cooled gas absorbs heat from the system it is cooling before the displacer forces the gas back to the warm end through the regenerator.

In an ideal system, with isothermal compression and expansion and a perfect regenerator, the process is reversible. Thus, the coefficient of performance COP for the ideal Stirling refrigerator is the same as the Carnot COP given by

$$COP_{carnot} = \frac{\dot{Q}_C}{\dot{W}_o} = \frac{T_c}{T_h - T_c} \qquad (1)$$

Where $\dot{Q}_C$ the net refrigeration power is, $\dot{W}_o$ is the power input, T_c is the cold temperature, and T_h is the hot temperature. The occurrence of T_c in the denominator arises from the PV power (proportional to T_c) recovered by the expansion process and used to help with the compression. Practical cryocoolers have COP values that range from about 1 to 25% of the Carnot value.

Stirling cycle consists of four thermodynamic processes acting on the working fluid: Points 1 to 2, Isothermal Expansion. Points 2 to 3, Constant Volume (known as isovolumetric or isochoric) heat removal. Isothermal Compression (Point 3 to 4), Points 4 to 1, Constant Volume (known as iso -volumetric or isochoric) heat addition [9].

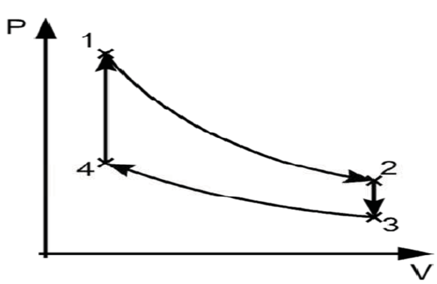

Fig. 3 : Stirling cycle

ii. *Pulse Tube Cryocoolers*

The displacer is eliminated. The proper gas motion in phase with the pressure is achieved by the use of an orifice and a reservoir volume to store the gas during a half cycle. The reservoir volume is large enough that negligible pressure oscillation occurs in it during the oscillating flow. The oscillating flow through the orifice separates the heating and cooling effects just as the displacer does for the Stirling and Gifford McMahon refrigerators. The orifice pulse tube refrigerator (OPTR) operates ideally with adiabatic compression and expansion in the pulse tube [10].

1. The four steps in the cycle are as follows.
2. The piston moves down to compress the gas (Helium) in the pulse tube. It flows through the orifice into the reservoir and exchanges heat with the ambient through the heat exchanger at the warm end of the pulse tube. The flow stops when the pressure in the pulse tube is reduced to the average pressure.
3. The piston moves up and expands the gas adiabatically in the pulse tube.
4. This cold, low pressure gas in the pulse tube is forced toward the cold end by the gas flow from the reservoir into the pulse tube through the orifice. As the cold gas flows through the heat exchanger at the cold end of the pulse tube it picks up heat from the object being cooled. The flow stops when the pressure in the pulse tube increases to the average pressure.
5. The cycle then repeats.

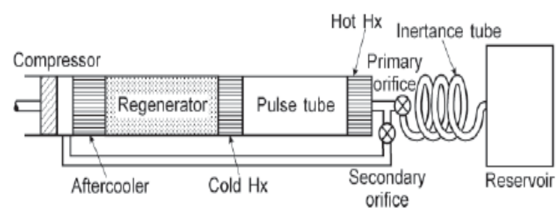

Fig. 4 : Schematic of pulse tube cryocooler with secondary orifice (double inlet) and inertance tube

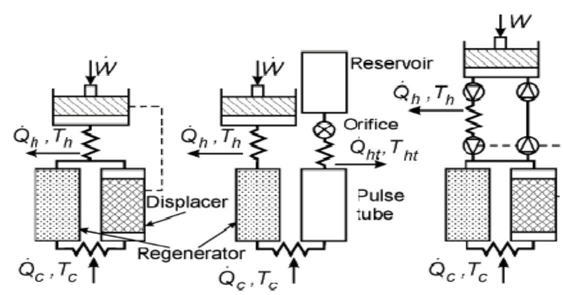

Fig.5 : Stirling cryocooler, pulse tube cryocooler and Gifford-McMahon cryocooler

IV. Pulse Tube Refrigerator Operation Principle

The operation principles of PTRs are very similar as conventional refrigeration systems. The methods of removing heat from the cold environment to the warm environment are somewhat different. The vapor compression cycle shown in Figure 6 operates in a steady flow fashion where heat is transported from the evaporator to the condenser by a constant and steady mass flow rate. The PTR relies on an oscillatory pressure wave in the system for transporting heat from the cold end heat exchanger to hot end heat exchanger.

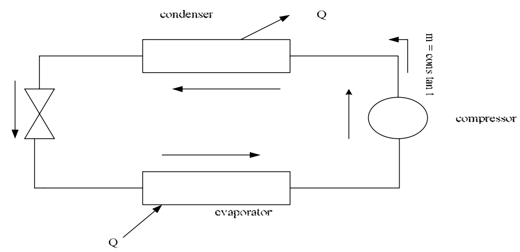

Fig. 6 : steady state flow of vapor compression cycle

In the pulse tube refrigerator the cooling actually occur in the oscillating pressure environment. The heat is absorbed and rejected at the two heat exchangers. It is a cyclic process.

Because PTR operates in steady periodic mode, the thermodynamic properties such as enthalpy flow $\dot{H}$, heat flow $\dot{Q}$ and power $\dot{W}$ are evaluated in the form of cyclic integrals. The appropriate instantaneous thermodynamic properties are integrated over the entire cycle and divided by the period of that cycle to obtain the cyclic averaged quantity of interest [11]. For example, the compressor power is evaluated from the following integration.

$$\dot{W} = f\oint P\frac{dv}{dt}dt = \frac{1}{\xi}\oint P(t)\dot{V}(t)dt \qquad (2)$$

Where f is frequency is period of the cycle, P and V, are instantaneous pressure and volume respectively. The average enthalpy flow over one cycle $\dot{H}$ and average heat flow rate $\dot{Q}$ are also calculated similarly.

V. Ptr Efficieny

In an ideal PTR the only loss is the irreversible expansion through the orifice. The irreversible entropy generation there is a result of lost work that otherwise could have been recovered and used to help with the compression [12]. All other components are assumed to be perfect, and the working fluid is assumed to be an ideal gas. The COP for this ideal PTR is given by

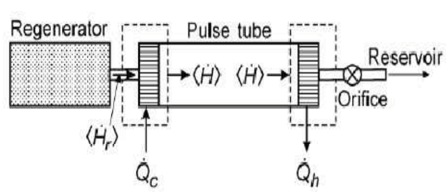

Fig. 7 : single stage orifice type pulse tube cryocooler with energy balance

$$COP_{carnot} = \frac{\dot{Q}_C}{\dot{W}_o} = \frac{T_c}{T_h - T_c}$$

$$COP_{ideal} = \frac{\dot{Q}_C}{\dot{W}_o} = \frac{(P_d V_c)}{(P_d V_h)} = \frac{T_c}{T_h} \qquad (3)$$

VI. Component Development

a) Expander

The expander assembly is the key cooling system component, enabling the actualization of a cryocooler with high efficiency, compact size, and low mass. The expander is a transducer that operates by creating an electrostatic force between two electrodes in a precision capacitor and allowing pressurized gas to separate the electrodes. The gas does work against the electrostatic force by separating the electrodes. This work is eventually dissipated as Joule heating in a warm load resistor. By doing work and removing it from the system, the expansion process can be carried out at nearly isentropic state and the dissipated energy provides an efficient means to reduce the gas temperature.

The expander is configured in an opposing piston arrangement and as gas is expanded on one side, the already cooled gas is expelled on the opposite side. In figure one side of the expander is being filled by opening a series of valves to the high pressure side of the system, figure the gas is expanded in the left side while the previously expanded and cooled gas on the right side is expelled to the low pressure side of the system.

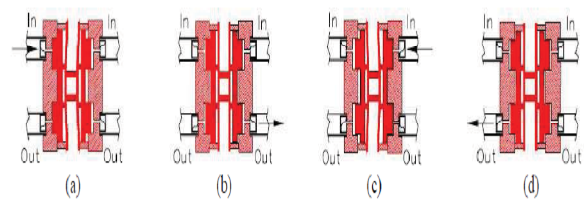

Fig .8 : Expander operation sequence

b) Compressor

An advanced oil free floating scroll compressor provides the DC flow required for application of the Brayton cycle in long life cryocooler systems. The floating scroll feature eliminates the prototypical scroll wear mechanisms by balancing the forces and resultant moment on the orbiting scroll while allowing the fixed Scroll to translate radically and axially, thereby minimizing contact forces between surfaces. Balance is achieved by configuring two orbiting scrolls mounted from a common base plate and mechanically driving the base plate from the outer edge or from a rigid central hub. Using this method, the forces can be reacted about the base plate producing no net off axis torque that can contribute to seal or wear.

To balance the axial forces that act on the scroll tips, an external gas pressurization scheme is employed. A pressurized gas volume is maintained external from the compression space on the backside of the fixed scroll to apply an axial force. The force on the fixed scroll will then just slightly exceed the separation force acting between the orbiting and fixed scrolls from the compressed gas. This applies a well controlled, nearly zero force to the tips, allowing sealing to occur without wear.

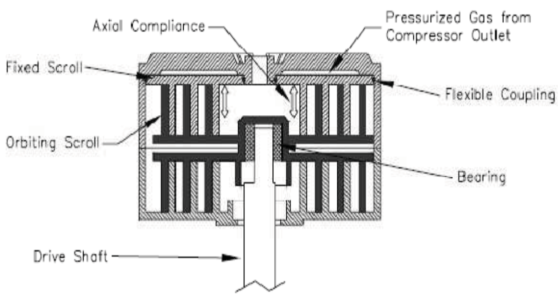

Fig. 9 : Floating scroll compressor

c) Heat Exchangers

Cryocooler uses a series of heat exchangers to achieve its thermodynamic efficiency, these include an after cooler to reject the heat generated in the compression process, recuperative counter flow heat exchanger between the high and low pressure gas streams, and cold end heat exchanger to interface with the element that are cooled. Effective heat exchange in each of the exchangers is paramount to achieving high system efficiency, but recuperate presents the largest challenge in terms of realizing a compact design that has high net effectiveness [13].

d) Regenerator

The regenerator is the most important component in pulse tube refrigerator. Its function is to absorb the heat from the incoming gas during the forward stroke, and deliver that heat back to the gas during the return stroke. Ideally, PTC regenerators with no pressure drop and a heat exchanger effectiveness of 100% are desired, in order to achieve the maximum enthalpy flow in the pulse tube. The performances of the real regenerators are of course far from ideal. Stainless steel wire screens are usually selected as the regenerator packing material, since they offer higher heat transfer areas, low pressure drop, high heat capacity, and low thermal conductivity.

e) Rotary Valve

It is used to switch high and low pressure from a helium compressor to the pulse tube system. The high and low pressure of helium compressor are connected to the rotary valve through the quick disconnect couplings. The rotary valve has a Rulon part which is made to rotate with the help of a synchronous motor against an aluminum block with predefined passages connecting the high and low pressures from the helium compressor [14]. The rotational frequency of the synchronous motor is controlled using an inverter drive.

The rotary valve has been designed to produce pressure wave in the frequency range from 1Hz to 3Hz. A typical design of rotary valve is shown in Fig 10.

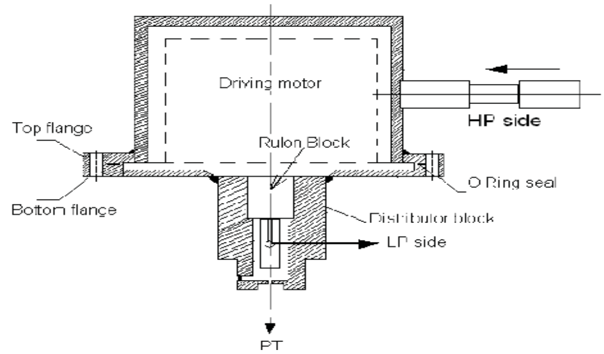

Fig. 10 : schematic diagram of rotary valve

VII. Review of Mathematical Analysis

There are three types of Pulse Tube Refrigerator Approximate Models, discuses in following.

a) First order of pump theory model

Gifford and Longs worth proposed a surface heat pumping theory to explain the performance of the basic pulse tube refrigerator after they have constructed a prototype of BPTR as shown in Figure 11. Consider a gas parcel in the pulse tube as shown in Figure. Suppose that in the beginning of the cycle the gas parcel at position X_1 has temperature T_1 and the temperature distribution of the wall is given as line 1-2. Consider the first half cycle where the pressure increases from the lowest to the highest. During this period, the gas parcel flows towards the closed end of the pulse tube to position X_2 undergoing an adiabatic process, hence its temperature increases to T_3. Since $T_3 \geq T_2$, therefore heat is rejected to the wall by the gas parcel until temperature of the gas parcel equals to that of the wall T_2. During the next half cycle, this gas parcel flows backward. This is an adiabatic expansion process where the temperature of the gas parcel decreases to T_4. Since $T_4 \leq T_1$, the gas parcel has refrigeration effect at the position X_1. This is so called surface heat pumping theory [15].

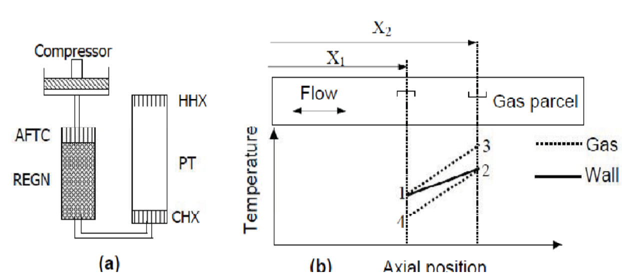

Fig. 11 : Surface heat pumping theory for BPTR

b) First order of enthalpy flow analysis model

Energy flow at various components for enthalpy flow analysis This figure demonstrates that the PTR's heat absorption and rejection occur at the cold heat exchanger (CHX) and the two hot heat exchangers; an after cooler (AFTC) and a hot end heat exchanger

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(HHX). HHX is equivalent to a condenser in a conventional vapor compression cycle, and CHX is equivalent to an evaporator. During the PTR operation, most of the heat generated due to compression is rejected through the after cooler. The rest of the energy that is not rejected through AFTC is carried through by the enthalpy flow $\dot{H}_{rg}$ in the regenerator. This can be seen in the component energy balance schematics shown in Fig 12. The regenerator enthalpy flow $\dot{H}_{rg}$, the additional refrigeration load $\dot{Q}_{refrig}$ and the heat flow representing all the losses, *loss* $\dot{Q}_{loss}$ (such as gas conduction, solid matrix conduction, and dispersion), are all absorbed at the CHX, therefore,

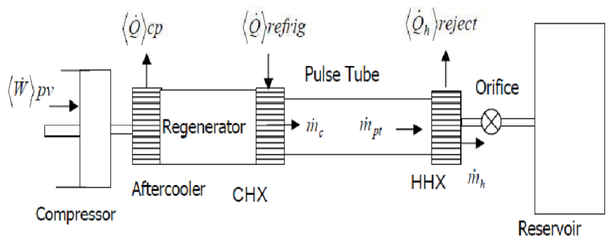

Fig. 12 : Enthalpy flow model

This enthalpy flow enters the pulse tube, and travels down the tube, reaches HHX and then part of this enthalpy is rejected to the environment. The remaining enthalpy which has not been rejected from HHX flows towards reservoir through orifice as shown in the energy balance figure. The average enthalpy flow over a cycle by assuming ideal gas flow is given by [16]

$$\dot{H}_{chx} = \frac{C_p}{\varsigma}\int_0^{\varsigma} \vec{m}.\vec{T}dt = \frac{C_p}{\varsigma}\int_0^{\varsigma} \dot{m}.T \cos dt \qquad (4)$$

Where ς the period of the cycle is, C_p is the heat capacity.

The pharos quantities $\vec{m}$ and $\vec{T}$ are mass flow rate and temperature respectively.

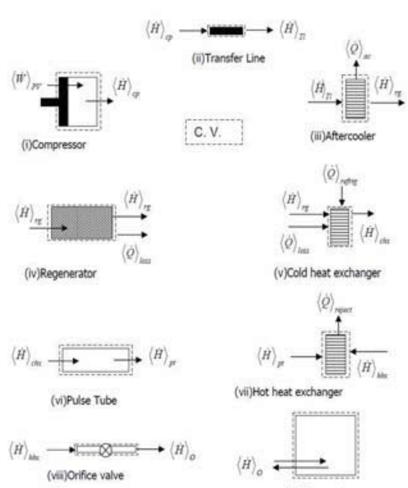

Fig .13 : Schematic diagrams for enthalpy flow model

According to the equation, if an oscillating mass flow rate $\dot{m}$ is in phase with the oscillating gas temperature T then a net enthalpy flow exists in the pulse tube flowing from the cold end to the warm end (i.e. $\dot{H}_{chx}$>10). Mass flow rate shown in right to right hand place shown in figure 14),On the other hand, if an oscillating mass flow rate $\dot{m}$ is out of phase with oscillating gas temperature T, then little or no enthalpy flow will exist in the pulse tube, which results in minimum cooling. Figure, depicts two examples of phase shift between gas temperature and mass flux.

The first example in Fig. 13 demonstrates a case where the mass flow rate and the temperature oscillations are about 90 degrees apart. In this circumstance, little or no enthalpy flow takes place. In fact, with temperature and the time mass flow rate being 90 degrees out of phase, one phase quantity will always be zero when the other one is at its peak. Thus, out of phase relationships tend to produce poor refrigeration due to minimum enthalpy flow in the pulse tube. On the other hand, if the mass flow rate and the temperature oscillations are in phase as illustrated in the second example (Fig.14), good enthalpy flow can exist in the pulse tube. Thus in phase and out phase are the two extreme conditions. In actual pulse tube there are exists same phase difference between the phase quantities [17].

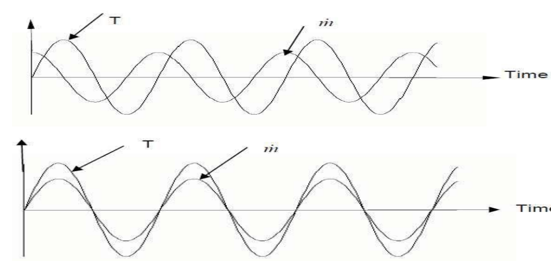

Fig. 14 : Mass and temperature analysis through phase distribution

For perfect regeneration without any loss,

$$\dot{H}_{rg} = 0,\ \dot{Q}_{loss} = 0$$

Refrigerating effect is obtained as,

$$\dot{Q}_{refrig} = \dot{H}_{chx} = \frac{C_p}{\varsigma}\int_0^{\varsigma} \dot{m}_c Tdt \qquad (5)$$

c) Second order of adiabatic model analysis

The working process of the pulse tube refrigeration system is very complex due to the unsteady, oscillating compressible gas flow, the porous media in regenerator, the presence of the orifice-reservoir, the double inlet valve etc. The cooling effect at cold end of the pulse tube occurs due to compression and expansion of the gas column lies somewhere between the adiabatic and isothermal processes, and may be assumed to be a polytrophic process. To understand the basic phenomenon responsible for the production of cold effect at the pulse tube section, two limiting cases adiabatic and isothermal processes involving ideal gas have been considered. Both these models are approximate models which are dealt separately.

The following assumption has been made with adiabatic behavior of the gas. The regenerator, the cold end and hot end heat exchangers have been assumed to be perfect. That means that the regenerator will always maintain a constant temperature gradient between its hot and cold ends at steady operation [18]. And heat addition at cold end heat exchanger and heat rejection at hot end heat exchanger of pulse tube occur at constant temperature at steady conditions.

1. The working fluid has been regarded as an ideal gas.
2. The gas flow in pulse tube has been assumed to be adiabatic in viscid flow with no length wise mixing or heat conduction.

(In the figure point 1 is compressor, 2 is after cooling, 3 is regenerator, 4 is cold end, 5 is pulse tube, 6 is hot end, 7 is orifice, DI valve and 9 is reservoir)

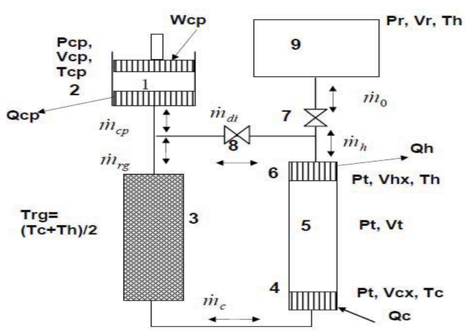

Fig. 15 : Schematic diagram of the pulse tube refrigerant with adiabatic model analysis

i. *Governing Equations*

The governing equations consist of continuity equation and energy equation.

$$\frac{\partial P}{\partial t}+v\frac{\partial P}{\partial x}-a^2\left(\frac{\partial P}{\partial t}+v\frac{\partial P}{\partial x}\right)=0 \qquad (6)$$

$a^2=\dfrac{\gamma p}{\rho}, v =$ The velocity of the gas along the tube

ii. *Adiabatic Compressor Modeling*

The pressure wave in the pulse tube is provided with a compressor directly coupled to the hot end of the regenerator. This design is more compact and more efficient than the valve compressor with gas distributor design [19]. The compressor cylinder has been assumed to be adiabatic in the analysis, since each of the compression and expansion processes occurs in such a short period of time that little heat exchange between the gas and the cylinder wall can be affected. The gas adiabatically compressed in the cylinder is assumed to be cooled to room temperature by the adjacent after cooler. The after cooler has been assumed to be perfect, so that the temperature of the gas leaving it is always equal to its wall temperature.

iii. *Change in Compressor Volume*

Sinusoidal variation has been taken for the compressor cylinder volume variation.

$$V_{cp}(t)=V_0+\frac{V_S}{2}[1+\sin(2\pi ft)] \qquad (7)$$

Where V_0 = clearance volume, V_S = stroke volume and f = frequency

Applying the first law of thermodynamics to the control volume drawn around the volume swept by the piston in the cylinder and Compressor pressure variation is expressed as

$$\frac{dp_{cp}}{dt}=[-\dot{m}c_pRT_{cp}-P_{cp}\frac{dV_{cp}}{dt}] \qquad (8)$$

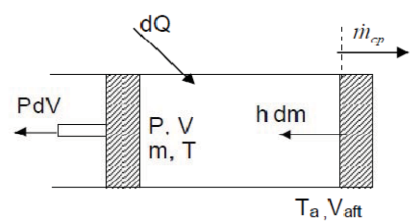

Fig.16 : control volume cylinder

iv. *Pressure Variation at the Pulse Tube*

Pulse tube pressure variation is a function of compressor pressure variation. So the pressure variation in the pulse tube can be derived in terms of compressor pressure variation along with various mass flow rate involved in the system.

The cold end mass flow rate equation derived earlier is

$$\dot{m}_c=\dot{m}_h\frac{T_H}{T_C}+\frac{V_t}{\gamma RT_c}\frac{dp_t}{dt} \qquad (9)$$

Where T_C and T_H are the temperatures at cold end and hot end respectively and R is a gas constant. In case of double inlet pulse tube refrigerator, the mass flow rate through the double inlet valve (DI) is due to the pressure difference between compressor and pulse tubes [20]. If DI valve mass flow rate is $\dot{m}_{di}$ and V_{dhx} is the void volume of the hot end heat exchanger and cold end mass flow rate is $\dot{m}_c$. The mass flow rate at the hot end of regenerator $\dot{m}_{rg}$, calculated by

$$\dot{m}_{rg}=\dot{m}_c+\frac{V_{dcx}}{RT_c}\frac{dP_t}{dt}+\frac{V_{drg}}{RT_{rg}}+\frac{dP_t}{dt} \qquad (10)$$

The cold end mass flow rate is given as,

$$\dot{m}_c=\left(\dot{m}_0-\dot{m}_{di}\right)\frac{T_h}{T_C}+\left(\frac{V_t}{\gamma}+V_{dhx}\right)\frac{1}{RT_c}\frac{dP_t}{dt} \qquad (11)$$

Compressor out let mass flow rate is given as:

$$\frac{dP_t}{dt}=\frac{R\left(\dot{m}c_p-\dot{m}_{di}\right)-R\left(\dot{m}_0-\dot{m}_{di}\right)\frac{T_h}{T_C}-\frac{V_{dac}}{T_h}\frac{d_{pcp}}{dt}}{\left[\frac{V_{drg}}{T_{rg}}+\frac{V_{dcx}}{T_c}+\frac{V_t}{\gamma T_c}+\frac{V_{dhx}}{T_c}\right]} \quad (12)$$

v. *Pressure Variation at the Reservoir*

Pressure variation at the reservoir is due to the mass flow through the orifice and it is given as:

$$\frac{dp_{cp}}{dt}=\frac{1}{V_r}\left(-\dot{m}_0 RT_h\right) \quad (13)$$

vi. *Mass Flow through Regenerator*

Mass flow in the regenerator has been evaluated through Argon's equation,

$$\dot{m}_{rg}=\frac{\rho\pi d_{rg}{}^2 d_h{}^2}{600L_{rg}\mu}\frac{\phi^3}{(1-\phi)}(P_{cp}-P_t) \quad (14)$$

Where ϕ the porosity of the porous medium is, ρ is the density of the fluid, d_h the hydraulic diameter, μ is the dynamic viscosity of the fluid and A_{rg} is the cross section area of the regenerator. Assuming $dx = L_{rg}$ (length of regenerator) and $dp = \Delta p = (P_{cp} - P_t)$

vii. *Mass Flow through Orifice*

Mass flow through the orifice has been assumed as a nozzle flow, calculated from well known formula for a nozzle with a correction factor [21].

$$\dot{m}_0=-C_d A_0\sqrt{2\frac{\gamma}{\gamma-1}\frac{P_t^2}{RT_h}\left[\left(\frac{P_t}{P_r}\right)^{\frac{2}{\gamma}}-\left(\frac{P_t}{P_r}\right)^{\frac{\gamma+1}{\gamma}}\right]} \quad (15)$$

Where $P_t < P_r$

$$\dot{m}_0=C_d A_0\sqrt{2\frac{\gamma}{\gamma-1}\frac{P_t^2}{RT_h}\left[\left(\frac{P_t}{P_r}\right)^{\frac{2}{\gamma}}-\left(\frac{P_t}{P_r}\right)^{\frac{\gamma+1}{\gamma}}\right]} \quad (16)$$

Where $P_t > P_r$

viii. *Mass Flow Rate through Double Inlet Valve*

Mass flow rate through double inlet valve has also been assumed as nozzle flow similar to that in the orifice. Here the mass flow occurs due to pressure differences between compressor and the pulse tube. Therefore, mass flow rate has been calculated as

$$\dot{m}_{di}=C_{di}A_{di}\sqrt{2\frac{\gamma}{\gamma-1}\frac{P_{cp}{}^2}{RT_h}\left[\left(\frac{P_t}{P_{cp}}\right)^{\frac{2}{\gamma}}-\left(\frac{P_t}{P_{cp}}\right)^{\frac{\gamma+1}{\gamma}}\right]} \quad (17)$$

Where $P_{cp} > P_t$

$$\dot{m}_{di}=-C_{di}A_{di}\sqrt{2\frac{\gamma}{\gamma-1}\frac{P_{cp}{}^2}{RT_h}\left[\left(\frac{P_t}{P_{cp}}\right)^{\frac{2}{\gamma}}-\left(\frac{P_t}{P_{cp}}\right)^{\frac{\gamma+1}{\gamma}}\right]} \quad (18)$$

Where $P_{cp} < P_t$

d) Second Order of Isothermal Model Analysis

In this model, the compression and expansion processes are considered as isothermal. It shows higher efficiency than the adiabatic or any other model of the pulse tube. For the purpose of analysis, a pulse tube refrigerator system is divided into a few subsystems, which are coupled to each other. Different researchers have used different schemes for dividing the full pulse tube refrigerator into subsystems. The pulse tube device has been divided into six open subsystems. Three of them exchange work, heat and mass with the surroundings (compressor, cold and hot volumes), while the others exchange mass only (regenerator, double inlet valve and orifice reservoir). It has been assumed that all heat exchanges are at constant temperature and that temperature of all subsystems exchanging heat is equal to those of the heat reservoirs. Another condition is that mechanical equilibrium is realized in each part of the device. These conditions lead to the model presented in Figure 17. The system described in the figure consists of six opened subsystems as(In the figure point 1 is compressor, 2 is after cooling, 3 is regenerator, 4 is cold end, 5 is hot end, 6 DI valve,7 is orifice and 8 is reservoir) [22],[23].

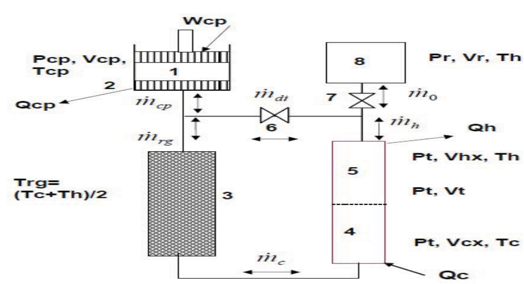

Fig. 17 : Schematic diagram of the pulse tube refrigerant with isothermal model analysis

1. Isothermal compressor at T_{cp}, exchanging mechanical work W_{cp} and heat q_{cp} with the surroundings at temperature T_h and mass ($m_{cp} = m_{rg} + m_{di}$) ; mass, *rg m* with the regenerator and mass, *di m* with the double inlet valve.
2. (Regenerator exchanging) mass m_{rg} at T_{cp} with the compressor. m_c At T_c with the pulse tube.
3. Cold volume in the pulse tube at T_c, exchanging heat $\dot{Q}_c$ with the surrounding (temperature T_c). Mass m_c with the regenerator, and mechanical work W_c with the hot volume.
4. Hot volume in the pulse tube at T_h , exchanging mechanical work W_c with the cold volume, heat q_h

with the surrounding at temperature T_{cp} and mass $(m_h = m_o - m_{di})$ with the reservoir via orifice and with the DI valve.

5. Adiabatic orifice and reservoir exchanging mass m_o with hot volume.
6. Double inlet valve exchanging mass m_{di} with compressor and hot volume [24], [25].

i. *Governing Equations*

Figure 18 shows a control volume which represents an isothermal variable volume.

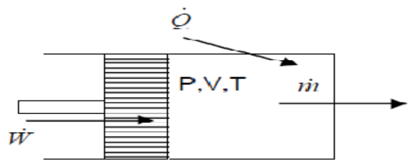

Fig. 18 : Control volume for isothermal model

$$\dot{Q} = P\frac{dv}{dt} - \dot{m}(pv) = P\frac{dv}{dt} - \dot{m}RT \tag{19}$$

Where, $\dot{w} = P\frac{dv}{dt}$

ii. *For Isothermal Compressor*

Applying the above sets of equations to the compressor

$$\frac{dp_{cp}}{dt} = \frac{1}{V_{cp}}\left(\dot{m}_{cp}RT_{cp} - p_{cp}\frac{dv_{cp}}{dt}\right) \tag{20}$$

$$\dot{Q} = P_{cp}\frac{dv_{cp}}{dt} + \dot{m}(pv) = P_{cp}\frac{dv_{cp}}{dt} + \dot{m}RT_{cp}$$

$$\dot{w} = -P\frac{dv_{cp}}{dt} \tag{21}$$

iii. *For Pulse Tube*

Similarly to that in compressor, the pulse tube flow has been assumed to be a piston like flow. In other words, the displacer of the Stirling or the GM cryocooler has been converted into a gas piston. The pulse tube has been divided into two distinct volumes, one for cold volume V_c and the other for the hot volume V_h at uniform temperature to ensure the reversibility of the model [26],[27].

$$\frac{dp_t}{dt} = \frac{1}{V_c}\left(\dot{m}_cRT_c - p_t\frac{dv_c}{dt}\right) \tag{22}$$

iv. *For Cold Volume*

$$\dot{Q} = P_t\frac{dv_c}{dt} - \dot{m}(pv) = P_t\frac{dv_c}{dt} - \dot{m}_cRT_c$$

$$\dot{w} = -P_t\frac{dv_c}{dt} \tag{23}$$

v. *For Hot Volume*

$$\frac{dp_t}{dt} = \frac{1}{V_h}\left(-\dot{m}_hRT_h - p_t\frac{dv_h}{dt}\right) \tag{24}$$

$$\dot{Q}_h = P_t\frac{dv_h}{dt} + \dot{m}_hRT_h \tag{25}$$

$$\dot{w} = -P_t\frac{dv_h}{dt}$$

The pressure variation in the pulse tube is the addition of two pressure variations in cold and hot volume.

$$\frac{dp_t}{dt} = \frac{1}{V_t}\left(\dot{m}_cRT_c - \dot{m}_hRT_h\right) \tag{26}$$

The fractional volume variation $X_t = \left(\frac{v_c}{v_t}\right)$ can be expressed by equating,

$$\dot{Q}_h = -P_tV_t\frac{dx_t}{dt} + \dot{m}_hRT_h \tag{27}$$

$$\dot{Q}_c = P_tV_t\frac{dx_t}{dt} - \dot{m}_cRT_c \tag{28}$$

vi. *Orifice and Reservoir*

For the reservoir, equations become,

$$\frac{dp_r}{dt} = \frac{1}{V_r}\left(-\dot{m}_0RT_h\right)$$

$$\dot{Q}_r = \dot{m}_0RT_h \quad , \dot{w} = o \tag{29}$$

e) *Third order of design data for Adiabatic and Isothermal models analysis*

Components	Parameters
Compressor	Dead volume V0 Swept volume Vs
Regenerator	Length L_{rg} Diameter d_{rg} Porosity Hydraulic diameter d_h
Pulse tube	Length L_t Diameter dt
Cold end block	Dead volume V_{dcx}
Hot end block	Dead volume V_{dhx}
Orifice	Diameter
DI valve	Diameter
Reservoir	Volume
Average pressure	Bar
Frequency	Hz
Cold end temperature	In temperature
Hot end temperature	In temperature
Helium gas at bar and Temperature	Dynamic viscosity, ρ, Cp, R, γ

f) Flow chart of the computer program for numerical simulation

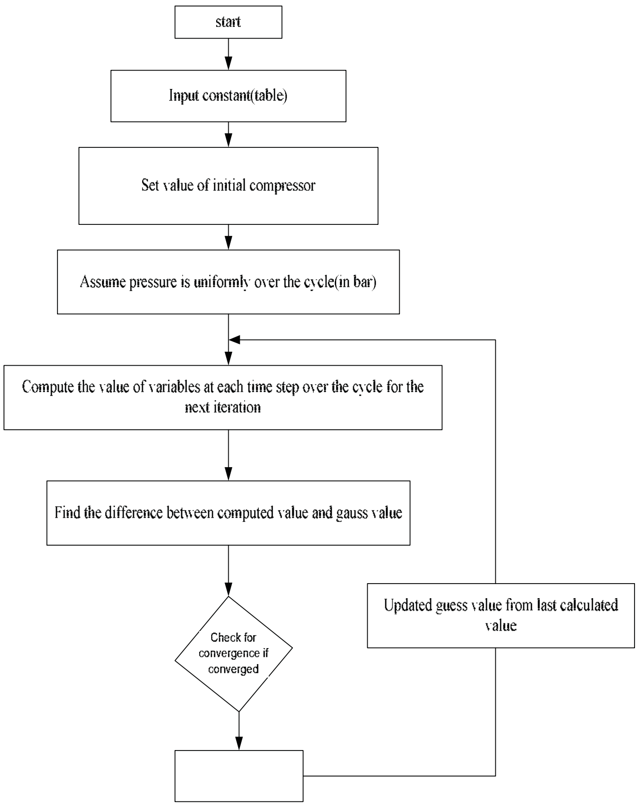

Fig. 19 : Flow chart of numerical simulation

VIII. Conclusion

In this study, first part is basic study of different types of cryocooler. Result is pulse tube type cryocooler is more reliable, no vibration etc. Second part, PTR efficiency method, flow properties, characteristic analysis and mathematical analysis use to find PTR different kind of equation to help for simulation techniques and various type of software such that fluent, CFD, and MATLAB etc. Mathematical analysis also use to find improved design and modification, it has now become the most efficient cryocooler for a given size. It is suitable for a wide variety of application from civilian to government to military and from ground equipment to space systems.

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A Novel Approach for the design of 2D Sharp Circularly Symmetric FIR Filters

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GJRE-F Classification : *FOR Code: 100508,100599*

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Manju Manuel [α] & Elizabeth Elias [σ]

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I. Introduction

Over the last few decades, the area of the design and implementation of 2D filters has attracted much attention of the researchers owing to its wide deployment in various areas like biomedical signal processing, seismic signal processing, radar and sonar procesing, genomic signal processing, satellite image processing etc(Williams et.al.,2007, Boudjelaba et.al.,2011). The various methods for the design of 2D linear phase FIR filter include frequency sampling, windowing, and frequency transformation(Lim,1990). Although the first two methods produce a better approximation of the response to the ideal one, the filter design involves large computational efforts especially for higher order filters. In contrast, the McClellan transformation has been bestowed with the features of reduced computation time and lesser implementation complexity. The higher order designs of the 2D zero phase filter can be done with much less computation efforts(Merserau,1980). The above features have contributed to lot of significant research performed in this area and very recently, this technique has been extended to the design of 3D filters(Mollova and Mecklenbrauker,2009).

Two dimensional high pass filters are extremely useful in detecting the sharp edges and boundaries of a given image. For instance, in weather monitoring, the discrete 2D high-pass filters are used to isolate the boundaries and edges of the weather data to make appropriate inferences and predictions(Feser and Storch,2005). Besides, high order high pass filter is shown to be a useful tool to evaluate and interpret meteorological data in finite areas in the work by Raymond,1989. The 2D high pass filter along with its complimentary lowpass filter has also been deployed in applications like the two channel quadrature mirror filter bank and the crossover network for image coding(Mitra and Yu,1986). There are several techniques for the design of 2D low pass filters(Lim,1990) and hence in most cases, the high pass filter is obtained using the complimentarity property(Mitra and Yu, 1986). Hence the direct design of high pass filters has not been addressed properly. In this paper, we propose a novel method for the direct design of 2D high pass filter. To this end, we use an approach similar to that is used in McClellan transformation using Chebyshev polynomial approximation for mapping the 1D filter into its 2D equivalent. But the striking difference is that the proposed transformation is used to derive a High Pass 2D filter directly from the 1D low pass prototype filter. Such an approach has not been reported in the literature so far. An advantage of using this approach is that this method preserves most of the characteristics of the 1D filter, especially the transition width and ripple characteristics. Once the high pass 2D filter is obtained, the complimentary low pass filter can be derived. This technique also avoids the drawback of the traditional McClellan transformation giving squarish contours at wide-band cutoff radii. The above transformations being multiplier-less, the number of multipliers needed for the realization of the 2D filter is decided by the order of the 1D FIR filter. Hence the number of multipliers of the 2D filter is fully decided by that of the 1D prototype filter.

In order to obtain a two dimensional filter whose performance specifications are close to an ideal 2D filter, the one dimensional prototype filter should be sharp. Even though 1D linear phase FIR filters are

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acclaimed for their inherent stability, computational complexity becomes exceedingly large as the transition width is reduced. To circumvent this drawback, we utilize the Frequency Response Masking (FRM) approach(Lim,1986) for the design of the sharp 1D prototype low pass filter and it results in sparse coefficients. FRM technique offers significant savings in the number of multipliers compared to the traditional mini-max design for the design of 1D filter. Due to the enormous computational saving possible for the design of an FRM filter, it has been extended to a variety of applications like array beam forming, FPGA, audio processing etc(Lim.et.al.,2007). Thus the 1D filter designed using FRM is converted into the 2D high filter using the proposed transformation. In many of the previous work using McClellan transformation, either optimization procedures or numerical methods were performed to obtain good circularity of the 2D radius. But in the very recent work by Liu and Tai, 2011, none of the above techniques were needed to obtain circular contours even at wideband. Instead a cascading term was added to the traditional McClellan transformation and two transformations namely T1 and T2 were proposed in the above paper. Since an analytical expression has been derived, the contour mapping problem is avoided in the T1 and T2 transformations. Our approach for the design of circular symmetric filters is based on the derivation of a totally new transformation instead of the mapping function in the traditional McClellan transformation. The contour mapping problem is avoided in our proposed method also. Hence the computation time is reduced drastically. In short, the sharp 2D filters designed using the proposed technique is better for high speed and low power applications while offering good performance close to the ideal one. This paper is organized as follows. Section II deals with an overview of frequency response masking approach. In Section III, the proposed transformation is briefed. The design of the 1D FRM filter is explained in Section IV A. Section IV B addresses the proposed design of the 2D circularly symmetric sharp filters. Simulation results are presented in Section V and conclusions are made in Section VI.

II. Overview of Frequency Response Masking

Frequency Response Masking is a much acclaimed technique for the design of an arbitrary bandwidth sharp FIR filter with reduced computational complexity. Efficient hardware implementation of the filter designed using FRM is possible due to the large number of zero valued coefficients. It consists of a prototype filter $H_a(z)$, complimentary filter $H_c(z)$, masking filter $H_{ma}(z)$ and a complimentary masking filter $H_{mc}(z)$. The complimentary filter $H_c(z^M)$ can be realized by substracting the output of the low pass filter $H_a(z^M)$ from the delay block $z^{-0.5(N_a-1)M}$, provided, the model filters are FIR in the nature. The block diagram of the FRM filter is shown in Fig. 1(Y.C.Lim, 1986).

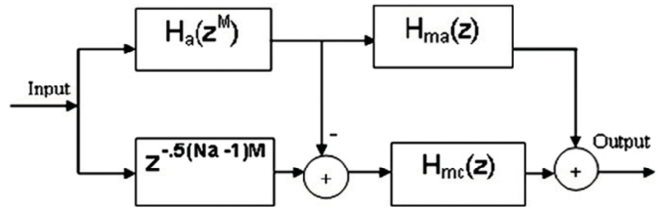

Figure.1 : Block Diagram of an FRM FIR filter.

The overall transfer function of the FRM filter can be written as follows

$$H(z) = H_a(z^M)H_{ma}(z) + H_c(z^M)H_{mc}(z) \quad (1)$$

where $H_c(z^M) = \left(z^{-0.5(N_a-1)M} - H_a(z^M)\right)$

The prototype filter, also called the band edge shaping filter, is interpolated by a factor M and therefore its transition width is reduced by a factor of M. The masking filters are used to retain the necessary spectrum repetitions for the formation of any arbitrary bandwidth filter under consideration.

III. Proposed Frequency Transformation

a) Derivation of the Proposed transformation

Generalised McClellan transformation converts 1-D linear phase filter $H(\Omega)$ into a 2-D linear phase filter $H(\omega_1,\omega_2)$ by means of the substitution of variables (McClellan, 1973).

$$H(\Omega) = \sum_{n=0}^{N} a(n)\cos\Omega n \quad (2)$$

where $a(n) = \begin{cases} h(0), for\ n = 0 \\ 2h(n), otherwise \end{cases}$

$h(n)$ corresponds to the 1D filter coefficients. Approximating $H(\Omega)$ using n-th order Chebyshev polynomial,

$$H(\Omega) = \sum_{n=0}^{N} a(n)\,T_n\,[\cos\Omega] \quad (3)$$

Now applying the 2D transformation by substitution of variables,

$$H(\omega_1,\omega_2) = \sum_{n=0}^{N} a(n)\,T_n\,[F(\omega_1,\omega_2)] \quad (4)$$

In the traditional McClellan transformation for designing the 2D filter with circular symmetry, the following expression was used(Liu and Tai, 2011).

$$\cos\Omega = F(\omega_1,\omega_2) = 2\cos^2(\omega_1/2)\cos^2(\omega_2/2) - 1 \quad (5)$$

Several investigations have been done in this direction to obtain low pass filters with different contours like circular, fan shaped, rectangular, diamond shaped etc.(Liu and Tai, 2011). But all of them were intended to obtain a 2D low pass filter. High pass filter was derived from it using the complimentarity property(Mitra and Yu,1986).

The advantage of using McClellan transformation technique compared to other filter design methods is that N number of multiplications are only needed to realize the 2D filter against N^2 number of multiplications in the other methods that use direct convolution if the filter is of size NxN. Making use of this advantage, in this work we propose a totally new transformation namely, H1 transformation for directly obtaining the 2D High pass filter from a 1D low-pass prototype filter.

This novel transformation is briefed below.

$$F(\omega_1,\omega_2)=(0.5\sin^2(\omega_1/2)+0.5\sin^2(\omega_2/2)-1)g(\omega_1,\omega_2) \quad (6)$$

where $g(\omega_1,\omega_2)= \cos^2(\omega_1/2)\cos^2(\omega_2/2)$

Linear Least square estimator has been used in deriving the transformation. The proof is given in Appendix 1. The features of the transformation are given as follows.

a. It is circularly symmetric at all radii.
b. It is quadrantly symmetric.
c. At all radii, the magnitude of $F(\omega_1, \omega_2)<1$.
d. The frequency mapping enables 1D prototype low-pass filter to become a 2D high pass filter.

The contour mapping of the proposed transformation is shown in Figure.2. A promising feature of the proposed design of the 2D filter is that the contour mapping problem is avoided in our work, since, an analytical expression is available for the proposed transformation. Hence, the 2D filter is obtained by the direct application of the proposed transformation. This calls for reduction in the computation time of the 2D filter.

b) Frequency Mapping of the proposed H1 Transformation

The relationship between the one dimensional frequency Ω and the two dimensional radius ω is derived below.

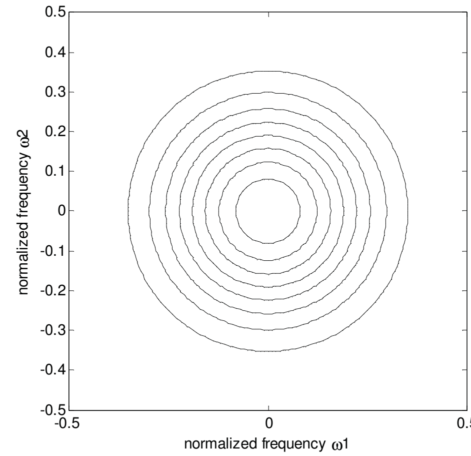

Fig.2 : Contour of the proposed transformation

Utilizing the quadrantal symmetry of the proposed transformation, let us consider only one direction in the 2D plane namely the ω_1 axis and make $\omega_1=\omega$ and $\omega=0$. This is done to obtain a one to one mapping between the frequencies in the 1D and 2D.

After applying the above condition, Equation.6. is modified as follows

$$\cos\Omega= F(\omega_1,\omega_2)= (0.5\sin^2(\omega/2)-1))\cos^2(\omega/2) \quad (7)$$

From Equation.7., it can be found that the 1D frequency can be obtained as given below.

$$\Omega=|\cos^{-1}((0.5\sin^2(\omega/2)-1))\cos^2(\omega/2))| \quad (8)$$

More clearly, the mapping is given by

$$\Omega=|\cos^{-1}(0.5\sin^2(\omega/2)\cos^2(\omega/2)- \cos^2(\omega/2))| \quad (9)$$

The figure showing the relation between the 1D and the 2D frequencies is given in Figure.3.

c) Realization of the proposed H1 transformation

In order to realize the transformation it has to be expressed in terms of z_1 and z_2.
Hence using the trigonometric identities

$$\cos\Omega= 2\cos^2(\omega/2) - 1 \text{ and}$$
$$\cos\Omega= 1 - 2\sin^2(\omega/2),$$

the transformation in Equation (6)can be rewritten as

$$F(\omega_1,\omega_2)=(1/4(1-\cos\omega_1)+1/4(1-\cos\omega_2)-1)(1/4(1+\cos\omega_1)(1+\cos\omega_2)) \quad (10)$$

Therefore the final transfer function of the transformation is given by

$$F(z_1,z_2)=(1/4(1-Q_1(z_1))+1/4(1-Q_2(z_2))-1)(1/4(1+Q_1(z_1))(1+Q_2(z_2))) \quad (11)$$

where $Q_1(z_1)=(z_1^{1}+z_1^{-1})/2$ and $Q_2(z_2)=(z_2^{1}+ z_2^{-1})/2$

After rearranging the terms, above expression can also be written as follows

$$F(z_1,z_2)=(-1/2(1+Q_1(z_1)+Q_2(z_2))(1/4(1+Q_1(z_1)(1+Q_2z_2))$$

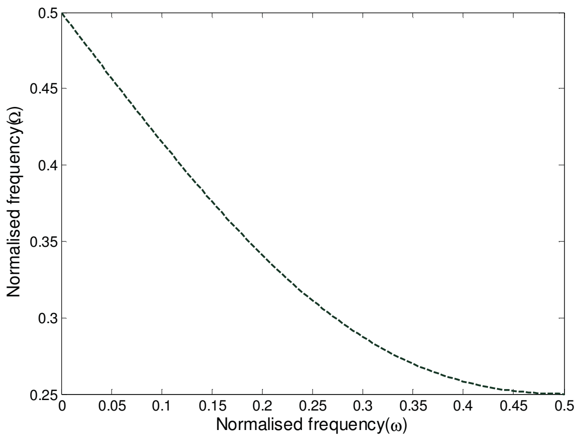

Fig.3 : Frequency Mapping of the proposed transformation

The block diagram of the proposed H1 transformation is shown in Figure.4. From the realization, it can be found that the transformation is totally multiplierless. Hence the total number of multipliers of the 2D filter is decided by the multipliers of the 1D filter. It is found that the computational complexity of the proposed H1 transformation, is least when compared to the T1, T2 and the McClellan transformation. The generalized McClellan transformation for obtaining circular contours as given in Kidambi, 1995 is used to make the comparison of complexity. The expression for McClellan transformation in Kidambi,1995 is given as follows.

$$\cos\Omega=0.5(\cos(\omega_1)+\cos(\omega_2))+t_{11}(\cos(\omega_1)\cos(\omega_2)-1) \quad (12)$$

The computational complexity of the transformation is found in terms of the number of adders and multipliers and the comparison is given in Table 1. MC refers to generalized McClellan transformation.

Table1 : Complexity comparison of various transformations

Complexity	H1	T1	T2	MC (Kidambi, 1995)
No of adders	8	10	9	6
No of multipliers	0	0	0	1

IV. Proposed Design of the Sharp 2D Circularly Symmetric Filters

A direct design of 2D high pass filter using frequency transformation has not been reported in the literature so far. All the work in this direction relied on the design of the 2D low-pass filter first, followed by finding out the complimentary high pass filter. The proposed design technique can be extended to the design of a 2D filter with the 1D prototype being any linear phase 1D filter. FRM has been used in our work to generate sharp filters with low complexity. Once the design specifications of the 2D High pass filter are obtained, using the relationship in Equation.9., the band edges of the 1D prototype low-pass filter are obtained. The design of the 1D prototype low pass filter using the FRM technique is briefed below.

a) Design of 1D FRM FIR Low Pass filter

The band edges of the various sub-filters of the FRM filter are obtained as given in the original work on FRM by Lim,1986. The optimal interpolation factor M is obtained in such a way that the total number of multipliers needed for the realization of the overall FRM filter is minimum. The sub-filters namely the model filter $H_a(z)$, masking filter $H_{ma}(z)$ and complimentary masking filter $H_{mc}(z)$ are designed as per the Remez Exchange algorithm. The conditions as outlined in the work by Lu .et.al., 2003 have been used to obtain a linear phase for the overall FRM FIR filter. As the objective of the work is to design a 2D high pass sharp filter with reduced complexity, FRM technique is used to obtain a sharp 1D filter with reduced complexity. The transformation preserves the properties of the 1D filter like the transition width, ripple characteristics etc.

b) Proposed Design of the 2D circularly symmetric Filters

A direct transformation is applied to the 1D prototype filter to obtain the sharp circularly symmetric High pass filter. A promising aspect of this design approach is that better circularity is obtained for all the contours when compared to the traditional McClellan Transformation. Also circularly symmetric 2D FRM based FIR filters have been reported for the narrow band case only.

Besides, a 2D circularly symmetric low pass filter can be obtained by the complimentarity property as given in the work by Mitra and Yu, 1986. Here, only one adder is needed in addition to number of adders needed for the realization of the high pass filter. This low pass filter has the advantage that it has lesser complexity compared to the T1 and T2 transformations proposed very recently for the design of circularly symmetric low pass filter(Liu and Tai,2011). The realization of the 2D filter is shown in Figure.5. This realization is quite similar to that used for the McClellan Transformation and the only difference being the use of a new $F(\omega_1, \omega_2)$. In this figure, a(n) corresponds to the coefficients of the 1D FRM FIR filter. A typical application of the complimentary pair of 2D filters is in the image coding system using the crossover network as pointed out by Mitra and Yu,1986.

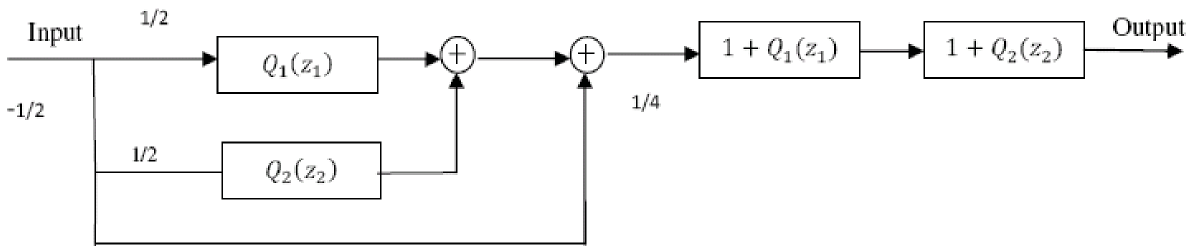

Fig.4 : Transfer Function of the proposed transformation $F(z_1,z_2)$

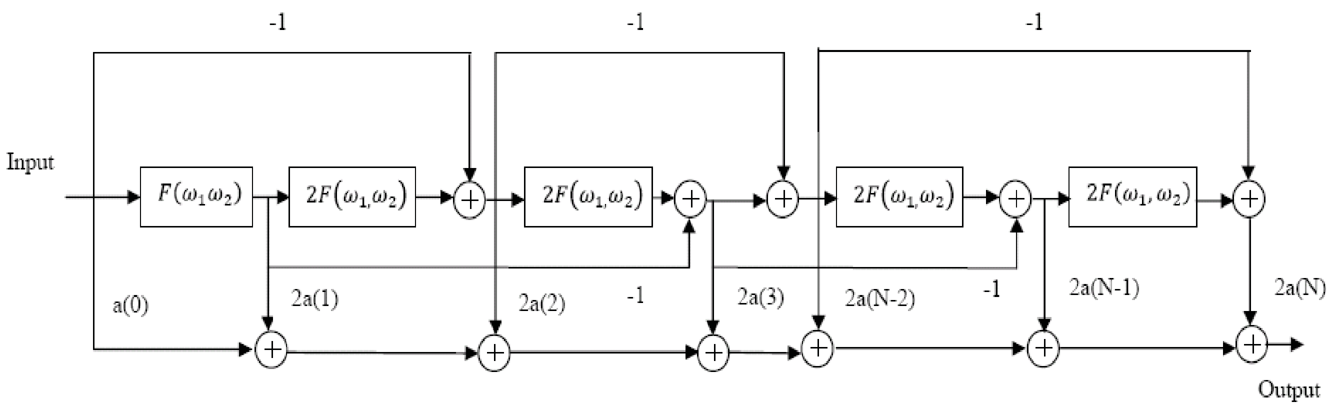

Fig.5 : Realization of the 2-D sharp High Pass filter

ω_r	.1π	.2 π	.3 π	.4 π	.5 π	.6 π	.7 π	.8 π	.9 π
$E(\omega_r)$	0.000051	0.000254	0.00085	0.0020	0.0037	0.0057	0.0073	0.007	0.006

Table 2 : Contour approximation error for varying 2D radius ω_r

In this paper, a new measure is proposed for evaluating the contour approximation error. Define C as the 1D frequency and ω_r as the 2D radius. The error measure $E(\omega_r)$is briefed as the L_2 norm of the error of approximation. It is given below.

$$E(\omega_r)=||\cos(\Omega)-F(\omega_r,\theta)||_2 \qquad (13)$$

The algorithm for finding the error is briefed below.

1. Fix the value of the 2D filter cutoff radius ω_r .
2. Vary theta for given cutoff radius and obtain $F(\omega_r, \theta)$.
3. For each radius, using inverse mapping, find the value of C
4. Compute $E(\omega_r)$ as per eqn.13.
5. Repeat the steps 2-4 for varying values of ω_r. The approximation error is tabulated in Table 2.

The computational complexity of a 2D high pass filter designed using the proposed H1 transformation is shown in Table.3. Here 2N+1 is the length of the 1D prototype filter. To derive the high pass filter, the complimentarity property was used. Here, only one adder is needed in addition to number of adders needed for the realization of the low pass filter. This high pass filter has the advantage that it has lesser complexity compared to the T1 and T2 transformation proposed very recently for the design of circularly symmetric low pass filter(Liu and Tai, 2011).

Complexity	H1	T1	T2	MC (Kidambi, 1995)
No. of adders	9N	11N+1	10N+1	7N+1
No. of multipliers	N+1	N+1	N+1	2N+1

Table 3 : Complexity comparison of implementation of 2D high pass filter.

V. Simulation Results

Simulation is done using MATAB 7.10.0 on a Dual Core Opteron processor. The proposed approach has been deployed to obtain typical FIR 2D high pass filter. From the high pass filter, the complimentary low pass filter can also be obtained. Two design examples have been illustrated to demonstrate the usefulness of the proposed approach. One wideband and one narrowband high pass filter example have been taken and the equivalent low-pass filter was found out in each case. The narrowband 2D high pass example is given as Case1. The wideband 2D high pass filter is given as Case 2.

Case 1

$$H(\omega_1,\omega_2,) = \begin{cases} \delta_s, & 0 \le \sqrt{{\omega_1}^2 + {\omega_2}^2} \le 0.6\pi \\ 1 \pm \delta_p, & 0.61\pi \le \sqrt{{\omega_1}^2 + {\omega_2}^2} \le \pi \end{cases}$$

$\delta_s = \delta_p = 0.01$.

The band-edges of the 1D prototype filter to be designed are found as $\Omega p = 0.5884\pi$, $\Omega s = 0.6101\pi$.

The magnitude response of the 1D prototype filter using FRM technique is shown in Fig.6. The magnitude response and contour of the 2-D circularly symmetric high pass filter for Case 1 are shown in Fig.7 and 8 respectively. The magnitude response and contour of the complimentary 2-D low pass filter for Case 1 are shown in Fig.9 and 10 respectively.

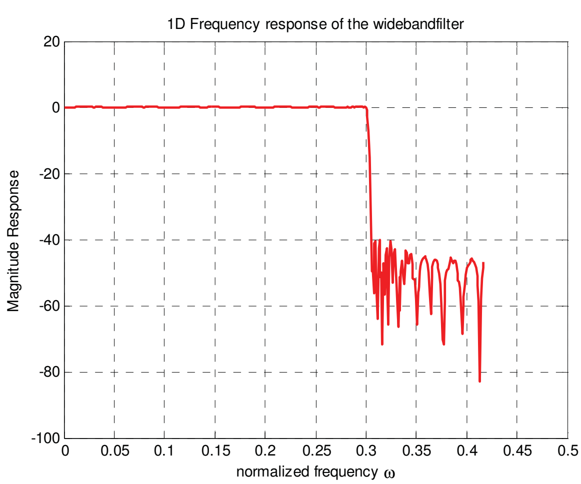

Fig.6 : Magnitude response of the 1D low pass prototype filter (Case 1)

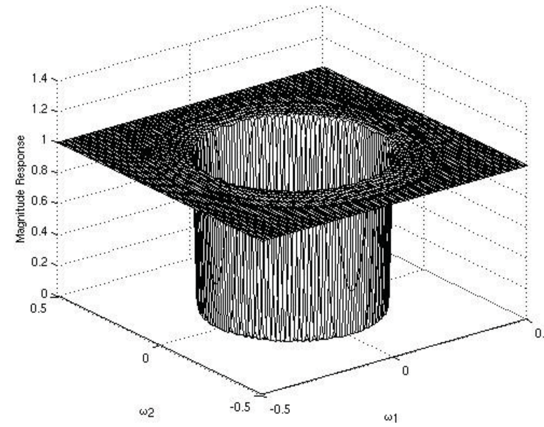

Fig.7 : Magnitude response of the 2D sharp high pass filter using the proposed H1 transformation (Case 1)

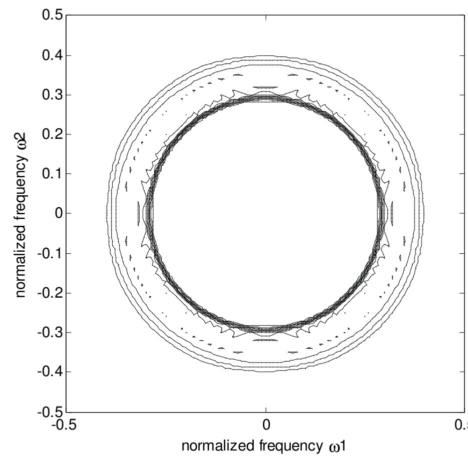

Fig.8 : Contour of the 2D sharp circularly symmetric high pass filter using the proposed H1 transformation (Case1)

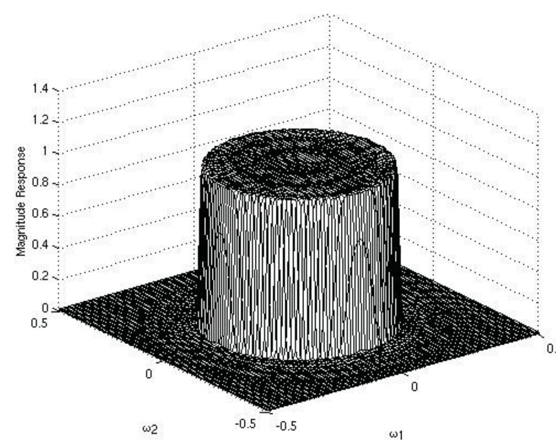

Fig. 9 : Magnitude response of the 2D complimentary low pass filter using the proposed H1 transformation (Case1)

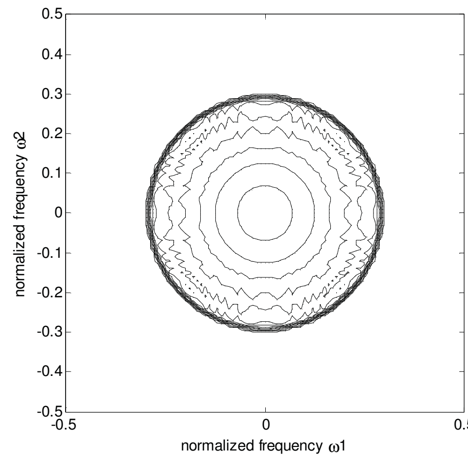

Fig. 10 : Contour of the sharp circularly symmetric 2D complimentary low pass filter using the proposed H1 transformation (Case1)

Case 2.

$$H(\omega_1, \omega_2,) = \begin{cases} \delta_s, & 0 \le \sqrt{{\omega_1}^2 + {\omega_2}^2} \le 0.2\pi \\ 1 \pm \delta_p, & 0.21\pi \le \sqrt{{\omega_1}^2 + {\omega_2}^2} \le \pi \end{cases}$$

$\delta_s = \delta_p = 0.01.$

The band-edges of the 1D prototype filter to be designed are found as $\Omega_p = 0.7933\pi$, $\Omega_s = 0.8086\pi$. The magnitude response and contour of the 2-D high pass filter for Case 2 is shown in Fig.11 and 12 respectively. Fig.13 shows the magnitude response of the complimentary low pass filter for Case 2.

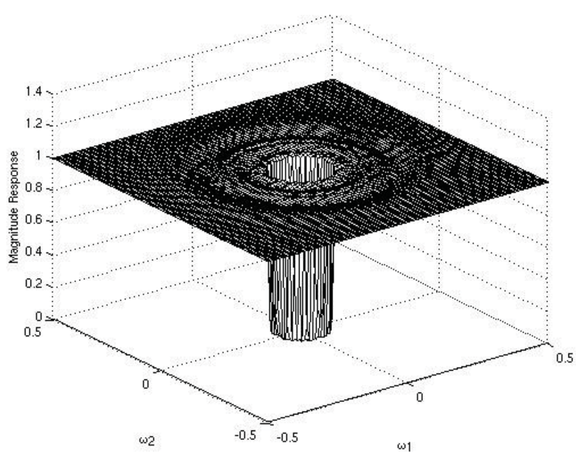

Fig.11 : Magnitude Response of the wide-band 2D sharp high pass filter using the proposed H1 transformation (Case 2)

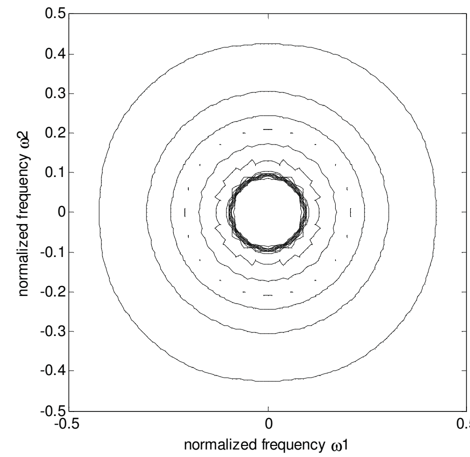

Fig.12 : Contour of the wide-band 2D circularly symmetric sharp high pass filter using the proposed H1 transformation (Case 2)

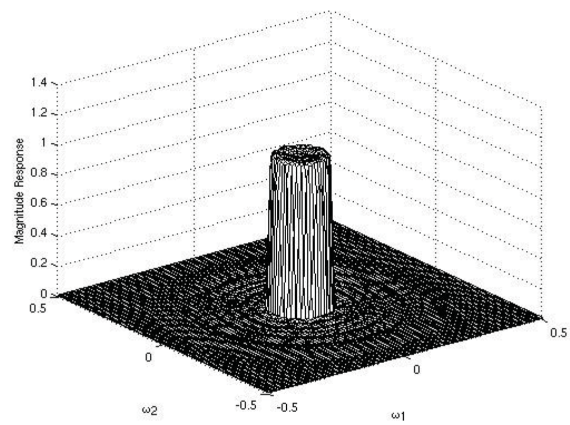

Fig.13 : Magnitude Response of the complimentary 2D sharp low pass filter using the proposed H1 transformation (Case 2)

VI. Conclusion

A novel and direct design approach for obtaining the 2D high pass, circularly symmetric, zero phase filter using the frequency transformation method has been presented. This transformation is then applied to a sharp 1D filter designed using Frequency Response Masking(FRM) technique. FRM is used to reduce the computational complexity of the sharp filter. From the 2D high pass filter, the complimentary low-pass filter can also be derived. The promising features of the proposed design technique is the reduced computation time and reduced complexity compared to the traditional methods. Good circularity is obtained at both narrowband and wideband radius of the 2D filter compared to McClellan transformation. Regarding the performance of the resulting 2D high-pass filter, the complexity is less compared to the recently proposed T1 and T2 transformations.

Appendix I

The proposed transformation for the direct design of 2D high pass filter has been found out using the Linear Least Square Estimator(Kay,1993).

Linear Least Square Estimator attempts to minimize the difference between the estimated and true parameter values. Here the unknown parameters are the coefficients of the basis functions being $\sin^2(\omega_1/2)$, $\sin^2(\omega_2/2)$ and 1 respectively. Consider the mapping given by

$$S=HU$$

S is the vector of dimension Nx1, H is an NxP matrix where N is the number of observations. P is the dimension of the vector U. X is the actual value. The least Square Estimate is found by minimizing the least squared error J(U) given by the following expression.

$$J(U)=(X-HU)^T(X-HU).$$

Setting the gradient of the the above =0, we get, the estimate of U as

$$\hat{U}=(H^T H)^{-1} H^T X$$

In the above method, the values of S are the known values of the expected contour defined at various values of the 2D frequencies(ω_1, ω_2). There are N observations. H is the matrix obtained by evaluating the values of the basis functions at various values of (ω_1, ω_2). In our work, we have N=20 and P=3. The estimated values of the coefficients of the constituents in the mapping are $\hat{u}_1=0.4848$, $\hat{u}_2=0.4848$, $\hat{u}_3=1$. In order to get a multiplier-les realization, and were taken to be 0.5. Hence the transformation is as follows.

$$0.5\sin^2(\omega_1/2) + 0.5\sin^2(\omega_2/2) - 1$$

But the contours of this transformation had noncircularity in the outermost radii. To correct this, a cascading term is added to the above transformation. Hence the resulting transformation was found to be as follows.

$$F(\omega_1,\omega_2)=(0.5\sin^2(\omega_1/2)+0.5\sin^2(\omega_2/2)-1)g(\omega_1,\omega_2)$$

Where $g(\omega_1,\omega_2)= \cos^2(\omega_1/2)\cos^2(\omega_2/2)$

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Computational Analysis of Return Loss for S-shape Microstrip Antenna

By Roli Nigam & Sanjay Kumar Srivastava

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Abstract - A survey of S-shape microstrip antenna elements is presented, with emphasis on theoretical and practical design techniques. Available substrate materials are reviewed along with the relation between dielectric constant tolerance and resonant frequency of microstrip patches. The S-shape antenna is first studied by a modal-expansion (cavity) technique and then is fully analyzed with wave equations. This paper presents the analysis of the electric & magnetic wave equation with the help of the numerical method. And results expressed that there is no energy loss in the propagation of wave.

Keywords : S-shape antenna, Return loss.

GJRE-F Classification : FOR Code: 100501

COMPUTATIONALANALYSISOFRETURNLOSSFORS-SHAPEMICROSTRIPANTENNA

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Computational Analysis of Return Loss for S-shape Microstrip Antenna

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Keywords : S-shape antenna, Return loss.

I. Introduction

Microstrip antennas are being increasingly used for aerospace applications because of their low weight, low volume and conformal nature. The most commonly used microstrip antennas are rectangular and circular disc antennas. However, other microstrip antennas are also being considered, depending on the application [1].

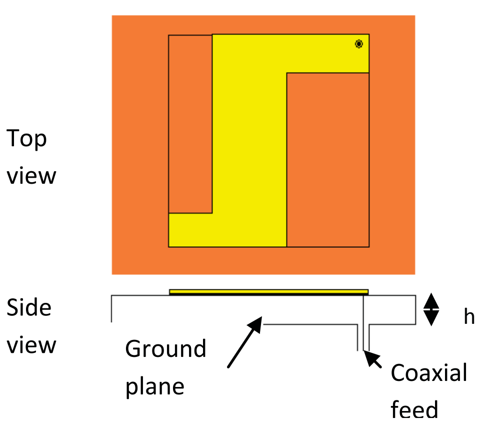

Fig.1 : Microstrip antenna configuration

In order to meet the requirement for mobile or personal communication systems, microstrip antennas with reduced size and broadband operation are of particular interest. Among various feeding mechanisms, the compact broadband microstrip antennas directly matched to a 50Ω coaxial line is also of importance, for its usefulness in integration with microwave integrated circuits. For this purpose, we present in this paper several related designs of microstrip antennas to broaden the operating bandwidth and reduce the overall size of the antenna. Here we discuss the S-shaped patch antenna. The S-shaped patch antenna reported here has a size about half that of the rectangular patch, with larger beamwidth but smaller bandwidth [2] shown in Fig.1.

II. Mathematical Analysis of S-Shape Antenna

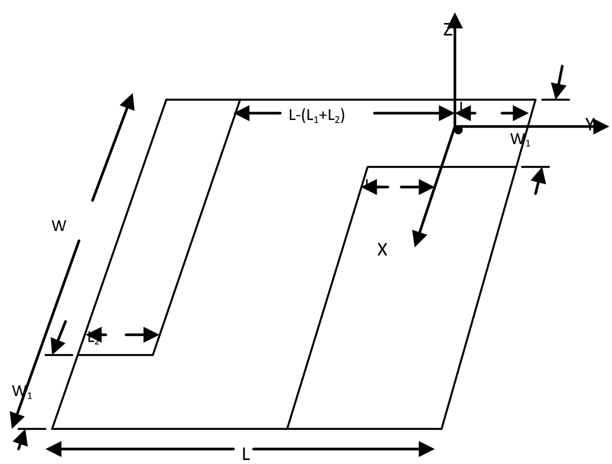

Fig. 2 : S-shape microstrip antenna with coaxil feed

Fig. 2. shows the geometry and co-ordinate system of the S-shape microstrip antenna. The width and length of the S-shape patch are W and L, respectively. The relative dielectric constant and the thickness of the substrate are ε_r and h, respectively. The antenna is excited at (x,y) by a coaxial feeder through the dielectric substrate. The electromagnetic fields in the x,y,z direction are denoted by (Ex , Ey, Ez) & (Hx, Hy, Hz) [3]. The electromagnetic fields within the cavity are divided into three regions by the boundaries through the feed point, as in Fig. 3. Moreover, as the antenna cavity is bounded on the side by the admittance wall, the electric field within the cavity is expanded in terms of the antenna parameters [5]. The electric field is expressed as

In region 1 (-W1/2 < X < +W1/2, -[L-(L1+L2)] < Y < L1, 0 < Z < h)

$$E_X = 0 \qquad (1)$$

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$$E_X = 0 \quad (1)$$

$$E_Y = E_0\cos\left(\frac{\pi}{[L_1 + (L-(L_1+L_2))]}y'\right) \quad (2)$$

$$E_Z = 0 \quad (3)$$

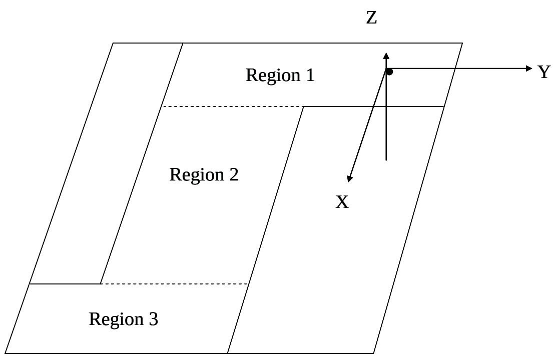

Fig. 3 : Geometry of S-shape antenna

In region 2 (W1/2 < X < [W1/2 + (W - 2W1)], -L3 < Y < (L3+{L-(L1+L2)}), 0 < Z < h]

$$E_X = EO\cos\left(\frac{\pi}{h}z'\right) \quad (4)$$

$$E_Y = 0 \quad (5)$$

$$E_Z = 0 \quad (6)$$

In region 3 ([W1/2 + (W-2W1) < X < W1/2 + (W-2W1) + W1], -L3 < Y < L3 + [L3-(L3 + L1)], 0 < Z < h)

$$E_X = 0 \quad (7)$$

$$E_Y = E_O\cos\left(\frac{\pi}{[L-L_3]}y'\right) \quad (8)$$

$$E_Z = 0 \quad (9)$$

And the magnetic field is expressed as

In region 1

$$H_X = H_O\cos\left(\frac{\pi}{[L-L_3]}y'\right) \quad (10)$$

$$H_Y = 0 \quad (11)$$

$$H_Z = 0 \quad (12)$$

Total electric field of S-shape antenna is presented by these equations which are expressed below [6]

$$E_X^{S-shape} = E_O\cos\left(\frac{\pi}{h}z'\right) \quad (13)$$

$$E_Y^{S-shape} = E_O\cos\left(\frac{\pi}{[L_1 + (L-(L_1+L_2))]}y'\right) + E_O\cos\left(\frac{\pi}{[L-L_3]}y'\right) \quad (14)$$

$$E_Z^{S-shape} = 0 \quad (15)$$

and magnetic field is

$$H_X^{S-shape} = H_O\cos\left(\frac{\pi}{[L_1 + (L-(L_1+L_2))]}y'\right) + H_O\cos\left(\frac{\pi}{[L-L_3]}y'\right) \quad (16)$$

$$H_Y^{S-shape} = 0 \quad (17)$$

$$H_Z^{S-shape} = 0 \quad (18)$$

III. Simulation Results and Discussion

As the design process goes the calculation of the parameters are done above and with the dimensions the S-shape patch antenna has been designed by the coaxial feeding techniques. Here, we take the coaxial feed technique in practice and the results are as shown below, A Matlab program has been compiled in order to produce the design and the responses for the S-shape microstrip patch antenna and its design. The table 1 below gives the possible parameters for the design of the microstrip patch antenna which will be used in the software for the results to examine. The width and the length of the patch have been rounded up to the close integer value.

Table 1 : Parameters used in software for the responses and simulations.

Parameter	Value
Dielectric constant of the substrate	4.2
Center Frequency	2.1GHz
Loss tangent	0.2

Width of the Patch	46.51mm
Length of the Patch	36.26mm
Height	1.6m
Zo	50Ω

The IE3D software has been compiled which gives us the following interface width (W) = 46.51mm length (L) = 36.26mm.

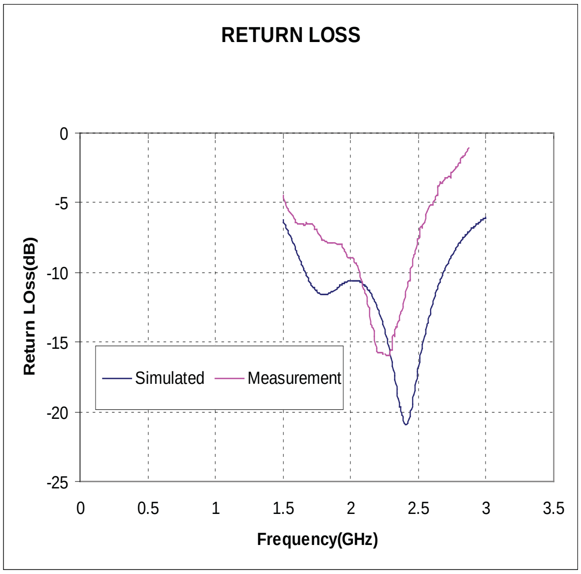

Fig.4 : Graph of the simulated & measured value

Graph of the simulated value & measured value are shown in the Fig.4. In these graph simulated and measured result are not closely matched. Simulated result is -23.34 dB at 2.4GHz resonating frequency [7] and measured result is -15.96db at 2.2GHz resonating frequency. At the operating frequency 2.1GHz the simulation result is -11.21db and the measured result is -10.73db. So that results presented that return loss are decreases without losses.

IV. Conclusion

The aim of this paper is to design a S-shape patch Microstrip antenna and to study the responses and electric and magnetic formulas are varies according to the boundry conditions. In this paper an antenna has been designed by the co-axial feeding technique. Having gone through the results it happened to be a bit difficult to decide the optimized design of the antenna, as there are different aspects that are involved in the design of patch antenna. It is good to see that the return loss has a negative value in all the cases which states that the losses are minimum during the transmission. In the design the Return Loss is -15.78dB in co-axial feed line technique for the simulated by the IE3D. But for the experimental design the Return Loss is -13.11dB in co-axial feed line.

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Temperature Control of a Hot Plate Using Microcontroller-Based PWM Technique

By Tabinder Akter, Md. Fazlul Huq , Farzana Alam, Md. Afzalur Rab & Dr.Md.Habibur Rahman

University of Dhaka

Abstract - The temperature of a hotplate used for prebaking, post baking, dehydration baking etc. changes with time. But, in baking purpose the temperature should remain constant. This work concerned with the designing a new and cost-effective microcontroller based PWM temperature controller to control the temperature of a hot plate. PWM controller has been selected because it has faster and accurate response than the conventional on-off controller. Here, the microcontroller senses the zero crossing of the line voltage and generates a pulse for every half-cycle. The width of the pulse is adjusted to change the firing angle of a TRIAC. The TRIAC is interfaced with the microcontroller circuit using an opto-coupler. The temperature of the hot plate can be increased or decreased using two push-button switches. The preset temperature will be saved in EPROM of microcontroller and displayed on an LCD display. The temperature of the hot plate is sensed by a thermocouple and fedback to the microcontroller. The microcontroller keeps the temperature constant by changing the firing angle of the TRIAC. The system has been constructed and tested successfully.

Keywords : Microcontroller, PWM controller, TRIAC, EPROM, Zero-crossing, Temperature Control.

GJRE-F Classification : FOR Code: 090601,090699

TEMPERATURECONTROLOFAHOTPLATEUSINGMICROCONTROLLER-BASEDPWMTECHNIQUE

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Temperature Control of a Hot Plate Using Microcontroller-Based PWM Technique

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General Terms : Hotplate, Opto-coupler, Step-down transformer, Comparator, and Peripheral interface controller (PIC).
Keywords : Microcontroller, PWM controller, TRIAC, EPROM, Zero-crossing, Temperature Control.

I. Introduction

Photolithography is very important in fabrication process [1]. During this process it is necessary to bake substances at different constant temperatures [2-4]. For baking purpose a hotplate is needed, whose temperature is not normally fixed [5]. Many manufacturers all over the world are supplying electronic equipment to control temperature of a hotplate but those equipments are very costly [6]. This work is intended to design low cost equipment with locally available materials.

The temperature of hotplate increases linearly and can't fix any temperature for a certain period of time. It is needed to fix the temperature of hotplate for a certain period of time. The work is intended to pre-bake or post-bake or hard-bake the Photo-resists spun onto the semiconductor wafers [1].

For a particular application of pre-baking the Positive Photo-resist spun onto 400-450 micron thick Silicon wafer of about 1 cm x 2 cm. A temperature of 100^0 C is needed to be fixed for 10 minutes or 30 minutes [7-9].

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The research conducted here is a combined hardware and software design process with an objective to develop a versatile control system. The work is divided in two sections: hardware section and software section.

II. Block Diagram of the System

The block diagram of the system is shown in figure 1. It consists of control unit, feedback unit, temperature heater, power control unit, PWM control, 220V line voltage, zero crossing detector.

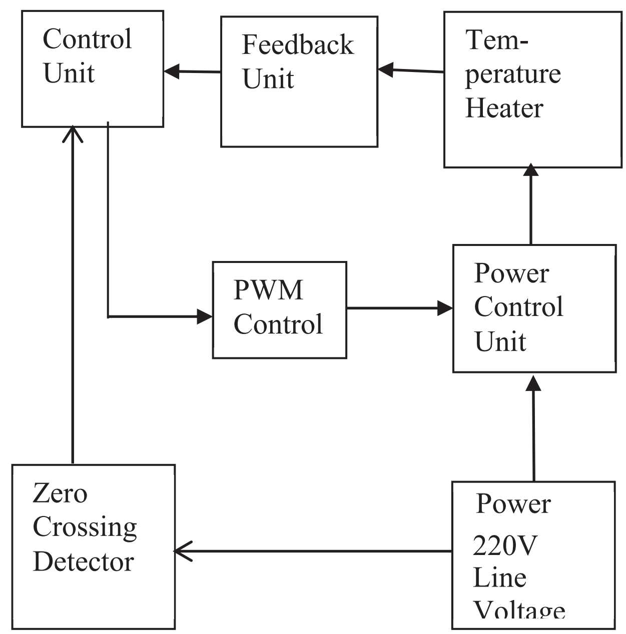

Figure 1 : Overview of the designed system.

III. Design & Development of the System

Hardware development

This section is divided into four subsections: Feedback unit, Power contro unit, control selection and PWM controller section.

a) Feedback Unit

The temperature of hotplate is displayed by an LCD which is connected with a 16F877 microcontroller. Hotplate converts temperature into voltage by a thermocouple. A high gained amplifier is used in the ADC input of the microcontroller. The hotplate temperature is fedback by this amplifier section. This section is important because the temperature sensed by a thermocouple is so small and need to amplify. The temperature is then compared with the preset value for adjusting with the target value.

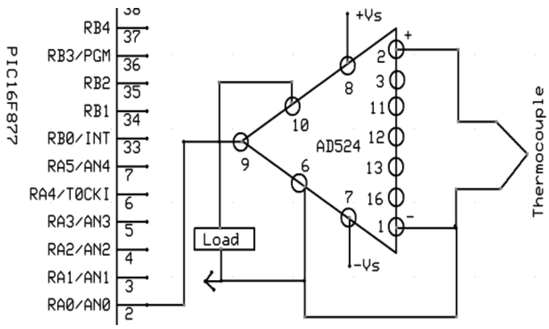

Figure 2 : Feedback Unit

b) Control unit

This unit works as the heart of the complete system. This unit is constructed using a microcontroller. It generates PWM signals and changes the width to keep the temperature constant [10]. It also synchronizes the PWM signal with the line voltage. To keep the plate temperature constant, it takes feedback from the plate. The plate temperature is sensed by a thermocouple and amplified by an instrumentation amplifier and applied to the microcontroller. The input voltage is then converted to digital signal by the ADC of the PIC16F877A and compared to the data stored in the EEPROM of the IC [11-13]. The LCD is connected with one of the output port of the microcontroller [14]. Switches have been connected with two input pins of an input port. The port was so programmed that it scans to check if the switch was pressed or not. By pressing the switches the set value of the temperature can be increased or decreased.

PIC16F877A have total 28 pins as shown in figure 3. Pin-1 ($\overline{\text{MCLR}}$ /VPP) of the IC is connected with VCC through a resistance [14]. This pin is the master reset pin. When the pin is connected with ground, the current program is reset and starts executing from the beginning of the program when again connected with V_{CC}.

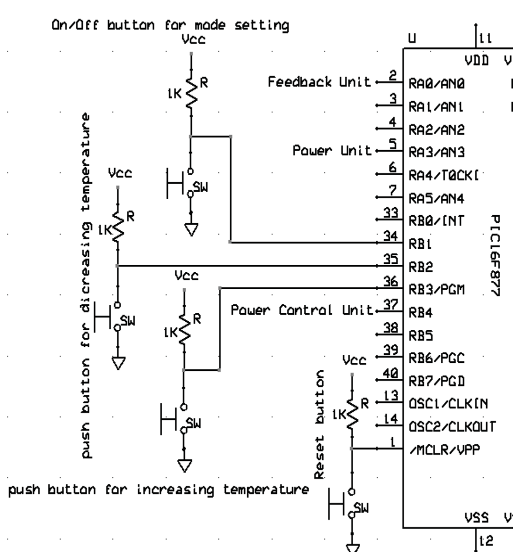

Figure. 3 : Control Unit.

c) Power Control Unit

To isolate the microcontroller based control system from the high power side an opto-coupler has been used to fire the TRIAC [14]. The main type of opto-coupler found in the local market is the type having an output DIAC or bilateral switch, and intended for use in driving a TRIAC or SCR. Examples of these are the MOC3020 and MOC3021. Here the output side of the opto-coupler is designed to be connected directly into the triggering circuit of the TRIAC.

As expected, the output of the DIAC is connected into the TRIAC gate triggering circuit in much the same way as a discrete DIAC. We need a filter/delay circuit before the DIAC (R1-2 and C1) and the usual snubber circuit across the TRIAC (Rs,Cs) to ensure correct triggering with inductive loads. Normally also need at least an RFI suppressor choke LRFI as well, plus a suitable capacitor across the load [14].

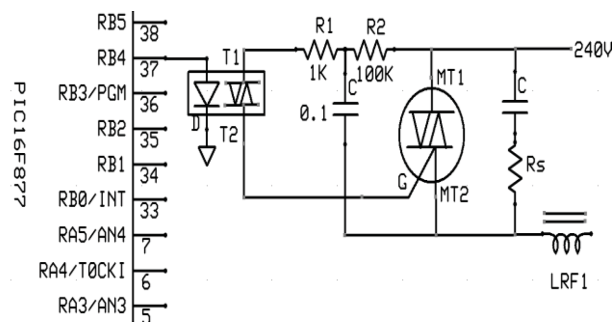

Figure. 4 : Power Control Unit.

d) PWM Controller

In this system, the line voltage has been lowered down by a step-down transformer and the sine-wave is directly given into a comparator of the microcontroller [15] which acts as a zero crossing detector. The comparator compares this signal with zero reference voltage. Whenever a transition is produced at the output of the comparator, the microcontroller begins a pulse whose width depends on the preset value of the temperature.

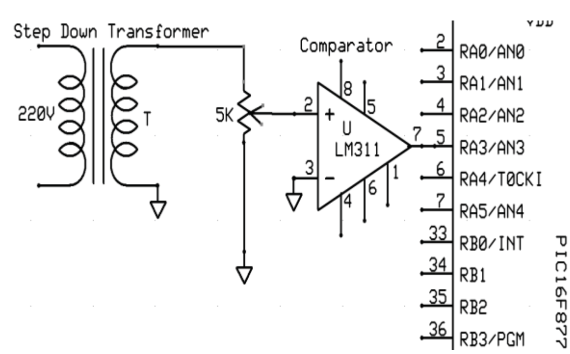

Figure. 5 : Power Unit.

IV. Software Development

The system is fully controlled by a program. The program has been written using FlowCode software. First, we set a temperature as shown in figure 6.

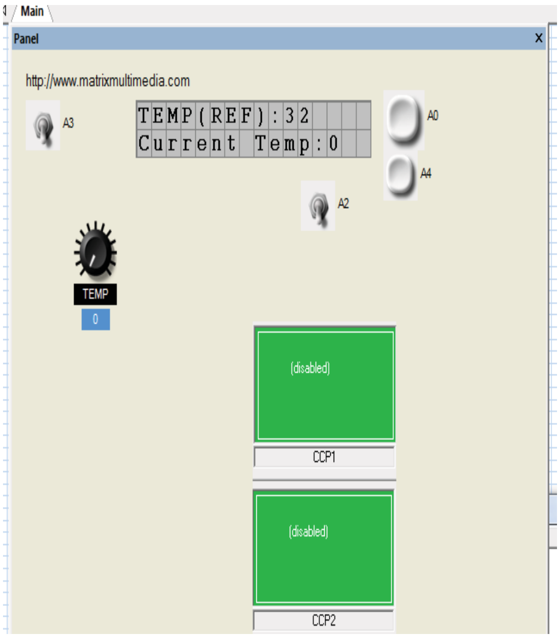

Figure. 6 : Output of the program

Then we must on the switch A3, A2 and also on the temperature button. It will start to work and set the hotplate's temperature. The output is shown in figure 7.

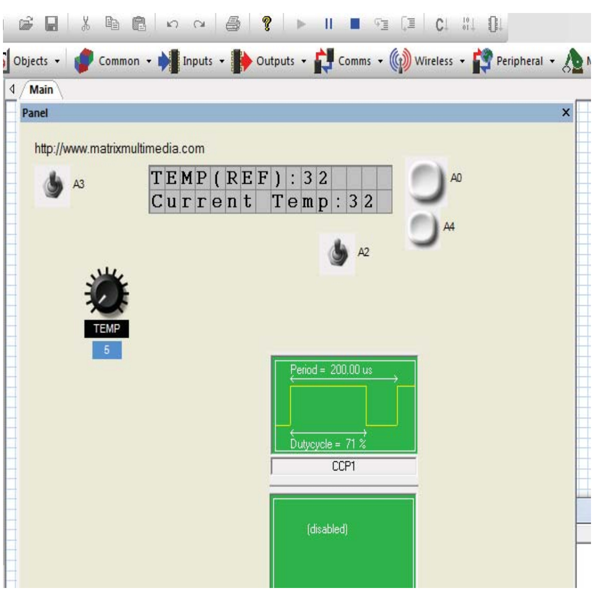

Figure. 7 : Output of the program (continued)

Here, hotplate temperature is controlled using duty cycle. For greater temperature it increases its duty cycle and for lower temperature it decreases its duty cycle for controlling the temperature.

V. Results & Discussion

a) Temperature Reading of a Hotplate

Our system is now ready to fix the temperature. We set our circuitry with the hotplate and control its temperature [3]. Our designed program is nice and work properly. The observed data of a hotplate for a set value of temperature is given below:

Data Table 1 : (For Preset Temp. 70°C)

TIME	TEMPERATURE
0.2	15
2	40
4	60
5	68.2
6	70
10	70

The graph associated with the table is shown below:

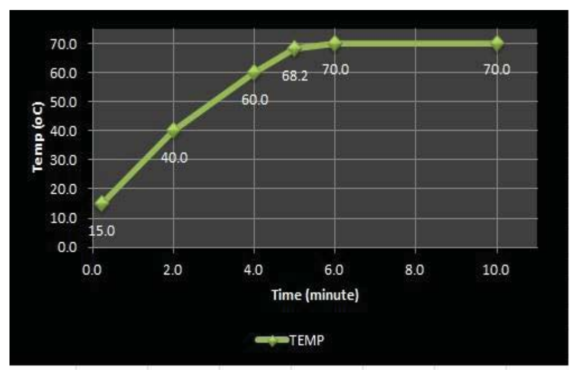

Figure. 8 : Graph for Preset temperature 70°C (Table 1).

It is evident that the temperature was increased from 15° C to 70° C for 5 minutes and after 5 minutes the temperature of the hotplate is fixed.

VI. Conclusion

The results of the performance study reveal that the developed system works properly with an excellent accuracy. One target of the system is to keep temperature of a device constant. A microcontroller chip 16F877A is used and it has a 256 bytes data EEPROM memory. So, this system can store up to 128 data. The LCD used in this system can display the stored data in EEPROM when the specific switch is pressed. Thus we can collect and analyze the data when necessary.

The other target of the system is to provide economic benefit to its users. Cheap and available components are used to design the system. The size of the system is small. Two ways - power and small size make the device portable. So, it can be easily carried and implemented anywhere.

The efficiency of the system is satisfactory. The A/D converter module has a high resolution and thus the accuracy of the system is nearly perfect.

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Comparative study of various Distributed Intrusion Detection Systems for WLAN

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Abstract - In any information system intrusions are the activities that damage the security and integrity of the system. In this paper we focus on wireless network, intrusions in wireless network (WLAN) and different Distributed Intrusion Detection Systems which are used to detect these attacks or intrusions. The rapid enhancement in wireless network has changed the level of network security. So, past of protecting the network with the firewalls are not sufficient to maintain network security in wireless local area network. There are different intrusion detection techniques which are used for identifying the various types of intrusions in wireless local area network. In this paper, we compare the various Distributed intrusion detection Systems used for detecting attacks in wireless network and also make a comparison table of these DIDS depending upon the performance. This comparison table will very helpful in designing better intrusion detection systems for detecting and preventing of vulnerabilities in wireless network.

Keywords : WLAN, Security, Intrusions, IDS, Intrusion Detection Systems, DIDS, Comparison Table.

GJRE-F Classification : FOR Code: C.2,C.2.1

COMPARATIVESTUDYOFVARIOUSDISTRIBUTEDINTRUSIONDETECTIONSYSTEMSFORWLAN

Strictly as per the compliance and regulations of:

Comparative study of various Distributed Intrusion Detection Systems for WLAN

Opinder Singh [α] & Dr. Jatinder Singh [σ]

Abstract - In any information system intrusions are the activities that damage the security and integrity of the system. In this paper we focus on wireless network, intrusions in wireless network (WLAN) and different Distributed Intrusion Detection Systems which are used to detect these attacks or intrusions. The rapid enhancement in wireless network has changed the level of network security. So, past of protecting the network with the firewalls are not sufficient to maintain network security in wireless local area network. There are different intrusion detection techniques which are used for identifying the various types of intrusions in wireless local area network. In this paper, we compare the various Distributed intrusion detection Systems used for detecting attacks in wireless network and also make a comparison table of these DIDS depending upon the performance. This comparison table will very helpful in designing better intrusion detection systems for detecting and preventing of vulnerabilities in wireless network.

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I. Introduction

Wireless networks are becoming so popular for many applications because they provide communication between different systems without predetermined infrastructure. Due to this flexibility new security risks are introduced in wireless network. The wireless network is dynamic in nature so there are number of challenges in maintaining security in wireless network. In wireless network there is need of defense schemes which are stronger, efficient and flexible. Intrusions in an information system are the processes or activities that damage the security policy of system. Intrusion detection is the process detecting and reporting unauthorized or unapproved network activity. It is used to identify intrusions or attacks against the system. Intrusion detection system (IDSs) collects and scrutinizes the data to recognize computer system and network intrusions or mishandlings. Conventional IDSs have been designed for wired systems and networks to identify intrusions or attacks. Of late, wireless network have been concentrated for employing the IDSs Constructed. Monitoring, analyzing user and system activities, identifying abnormal network activities and detecting policy violations for WLANs are the functions of these wireless IDSs. There are a lot of chances of attacks in WLANs due to dynamic topology, absence of infrastructure and centralized administration. Wireless IDSs collect all local wireless transmissions and rely either on predefined signatures [1] or on anomalies in the traffic [3] to produce alerts or alarms. In this paper we focus on different types of attack in wireless network, various distributed intrusion detection systems, research achievements in DIDSs fields and their comparison.

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II. Vulnerabilities in Wireless Lan

In wired network data travel from one place to another over a dedicated physical line that is private, but in WLAN data travel from one place to another over a shared space which is not private. It means there are more chances of vulnerabilities in wireless networks as compare to wired networks. Wireless networks have characteristics like dynamic topology, absence of centralized administration and low protection of nodes. Due to dynamic topology nature of wireless network there is no boundary of wireless network, so old methods like firewall protection are not applicable for security in WLAN. Different types of vulnerabilities in WLAN are:

a. **Due to lack of infrastructure:** In wireless networks there is no fixed infrastructure which makes different security mechanism inapplicable like certification, firewall and cryptography.

b. **Vulnerability due to channels:** In wireless network fake messages can easily be injected without making physical connection with the network.

c. **Dynamic topology:** In wireless networks dynamic topology is used which require sophisticated routing protocols. Problem arises due to mobility of devices. It is very difficult to track a misbehaving device in wireless network which generate wrong routing information.

d. **Vulnerability due to nodes:** In wireless network it is not possible to protect the different nodes physically. That is why these nodes can easily be captured by an attacker.

III. Various Type of Intrusions in Wlan

Webster's dictionary defines an intrusion as "the act of thrusting in or entering into a place or state without invitation, right or welcome", or an intrusion is an active sequence of related events that deliberately try to cause harm such as rendering system unusable, accessing unauthorized information or manipulating information. There are different types of Attacks in WLANs [4] which are:

a) Packet Dropping

Packet dropping is the type of attack in which some nodes drop data packets that are forwarded to another node and violate the operation of network. Packet Dropping attacks are further of two types.

a. Black Hole Attack: It is a type of attack in which attacker or misbehaving node drops all data packets.

b. Gray Hole Attack: It is a type of attack in which misbehaving node or attackers selectively drop data packets.

b) Worm Hole

It is kinds of attack in which a tunnel is created between some nodes that utilize secretly transmit packets.

c) Denial of Service

In this type of attack nodes are blocked from sending and receiving packets to their destinations.

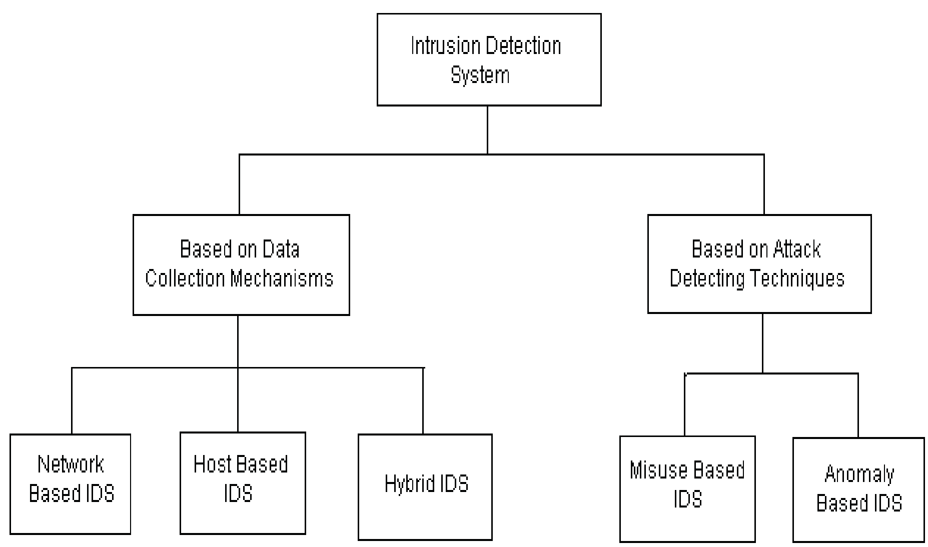

Figure1 : Classification of Intrusion Detection System.

d) Routing Loop

In this type of attack a loop is introduced in the normal path that violates the normal behavior of the network.

e) Delay Packet Transmission

In this type of attack an attacker nodes can transmit their own packets by delaying other's packets.

f) Fabricated route message

In this type of attack route messages are injected into the network that contains the malicious contents.

IV. Classification of Intrusion Detection System

Intrusion Detection Systems (IDSs) are the software designed for detecting, blocking and reporting unauthorized activities in computer networks. An Intrusion Detection System (IDS) can be categorized into two different forms according to data collection mechanisms and attack detecting techniques [4] as shown in figure 1.

a) Based on Data Collection Mechanism

An IDS can be categorized into three types [6] according to the data collection method: Network Based, Host Based, Hybrid intrusion detection system. Network based intrusion detection system reside on a separate system from where it watches the network traffic, looks for indications of attacks that traverse the portion of the network. Host based intrusion detection system resides on a particular host and looks for the indications of attacks on that host. Hybrid intrusion detection system has both the functionality of Network based and Host based intrusion detection system.

i. *Network Based IDS*

Network Based IDS (NIDS) exists as a software process on a dedicated hardware. The NIDS places the network interface card on the system into promiscuous mode, i.e. the card passes all traffic on the network to

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the NIDS software. The traffic is then analyzed according to a set of rules and attack signatures to determine if it is traffic of interest. If it is, an event is generated. Its attack recognition module uses four common techniques to recognize an attack signature:

- Pattern, expression or byte code matching,
- Frequency or threshold crossing
- Correlation of lesser events
- Statistical anomaly detection

Once an attack has been detected, the IDS' response module provides a variety of options to notify, alert and take action in response to the attack. Problem with NIDS is that it has high false positive rate. Another drawback is that in NIDS there is no central point to monitor whole N/W. So, it is not good for adhoc network.

ii. *Host-Based IDS*

HIDS exists as a software process on a system. HIDS examines log entries for specific information. Periodically, the HIDS process looks for new log entries and matches them up to pre-configured rules. If a log entry matches a rule, the HIDS will alarm. Today's host-based intrusion detection systems remain a powerful tool for understanding previous attacks and determining proper methods to defeat their future application. Host-based IDS still use audit logs, but they are much more automated, having evolved sophisticated and responsive detection techniques.

iii. *Hybrid IDS*

Hybrid intrusion detection system is an IDS which combine the functionality of network based sensor technology with host based agent that is capable of analyzing the network traffic only addressed to specific host where agent of hybrid IDS is installed [8].

b) *Based on Detection Techniques*

An intrusion detection system can be categorized into two different forms based on detection techniques: Signature or Misuse based and Anomaly based intrusion detection system.

i. *Signature or Misuse based IDS*

Misuse detection attempts to model abnormal behavior or signatures of known attacks. It is based on the assumption that all intrusions or attacks leave their signatures that can be detected[9,10]. Any occurrence of which clearly indicates system abuse. For Example, an HTTP request referring to the cmd.exe file may indicate an attack.

ii. *Anomaly based IDS*

Anomaly based IDS attempts to model normal behavior. Events that violate this model are considered to be suspicious. For Example, a normally passive public web server attempting to open connections to a large number of addresses may be indicative of a worm infection.

V. Various Architecture of Intrusion Detection Systems

Depending upon the infrastructure the wireless network can be divided into two different forms either flat or multi-layer. The best architecture of IDS for a wireless network depends upon the infrastructure of that network. The different types of IDS architecture are:

a) *Standalone Architecture*

In this type of architecture Intrusion Detection System (IDS) runs on each system to find out intrusions independently. In standalone architecture there is no data exchange and no cooperation among IDSs on the network. This architecture is more appropriate for network with flat infrastructure than for network with multilayered infrastructure [13].

b) *Distributed and Collaborative Architecture*

In this type of architecture every node in wireless network takes part in intrusion detection process with the help of IDS agent running on the different nodes. In distributed and collaborative architecture IDS agent is responsible for collecting and detecting the local events and data to find out different intrusions or attacks .After identifying the intrusion IDS give response at the same time [14].

c) *Hierarchical Architecture*

This architecture is the improved version of distributed and collaborative architecture. Hierarchical architecture is well suited for infrastructure of multi-layered network. In multi layered infrastructure network is divided into clusters and cluster heads in this type of infrastructure act as control points in the same way as routers, switches and gates in wired network [15].

d) *Architecture based on mobile agent*

In this type of IDS architecture mobile agents are used to perform required task on different nodes in wireless network. In mobile agent based architecture distribution of attack detection tasks are possible. It is very best method of using mobile agents [16, 18] for detecting intrusions.

VI. Literature Review of Various Distributed Intrusion Detection Systems (DIDS)

In 2002 Kachirski and Guha proposed an algorithm for Distributed Intrusion Detection System (DIDS)[15].This IDS is based on mobile agent technology. It is a multi-sensor IDS. In this IDS is divided into three different modules. Each of these module act as a mobile agent with some functionality like monitoring, initiating response and decision making. In this IDS functional tasks are divided into different categories and each task is assigned to different mobile agents. In this way workload is divided among different agents. This characteristic is good for wireless network.

Kachirski and goha also represent the hierarchical structure of different agents which is shown in figure 2.

Different functional tasks are performed by different agents like:

Monitoring: This type of task is performed by monitoring agent. There are two types of monitoring which is done by agents which are Network Monitoring and Host monitoring.

a) Host Monitoring

This task is performed by a host based monitor agent who hosts user activity sensors and system level sensors on every node for monitoring within node.

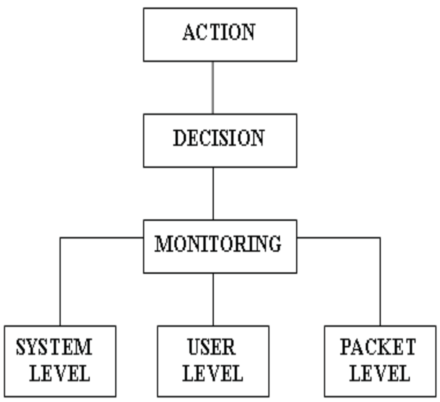

Figure 2 : Mobile Agent Architecture using different Layers.

b) Network Monitoring

In network monitoring sensors only runs on few selected nodes for monitoring at packet level to check whether packets are going through the network within their radio ranges or not.

Action: In this task each node acts as an action agent. When host based monitoring agent detects any unusual activity on host node then action agent gives response by blocking some user from the network or by terminating the task or process.

Decision: Every node in the network will decide about the attacks or intrusions threat level on the basis of host node. Some nodes in the network will gather the information regarding the intrusion and collectively make decisions for network level intrusions. In some cases when local detection agent can not able to take a decision due to some unsatisfactory evidence then it reports to decision agent for investigation .This is performed by considering packet monitoring results that are obtained from network monitoring sensor running locally. If the decision agent finds out that some node is creating intrusions in the network then action module carry out the response from that node. The wireless network is divided into different clusters with single cluster head for each. The purpose of this cluster head is to monitor the packets in cluster. It captures and investigates those packets which have their originators in same cluster. It means that decision agent and network monitoring agent both run on the cluster head. In this IDS decision agent makes decision from the information gathered by network monitoring sensor. Other nodes have no effect on decision made by decision agent. In this way attacks or intrusions can be prevented in wireless network.

In 2003 Y. Huang proposed a Cooperative and distributed intrusion detection system for wireless networks [14,19]. The architecture of intrusion detection system is divided into six different modules as shown in figure 3.

All the six modules as shown in figure work in systematic way. First of all the local data collection module accepts real time audit data. This audit data consist of user and system activities within radio range. This data is then transfer to the local detection engine for analyzing purpose. If an anomaly with strong evidence is detected by local detection engine then the IDS agent determine that the system is under attack. After detecting attack in the system it initiate a response with the help of local or global response module. Choice of response module depends upon the intrusion type, certainty of evidence and type of protocols. If an intrusion is detected without sufficient evidence then IDS

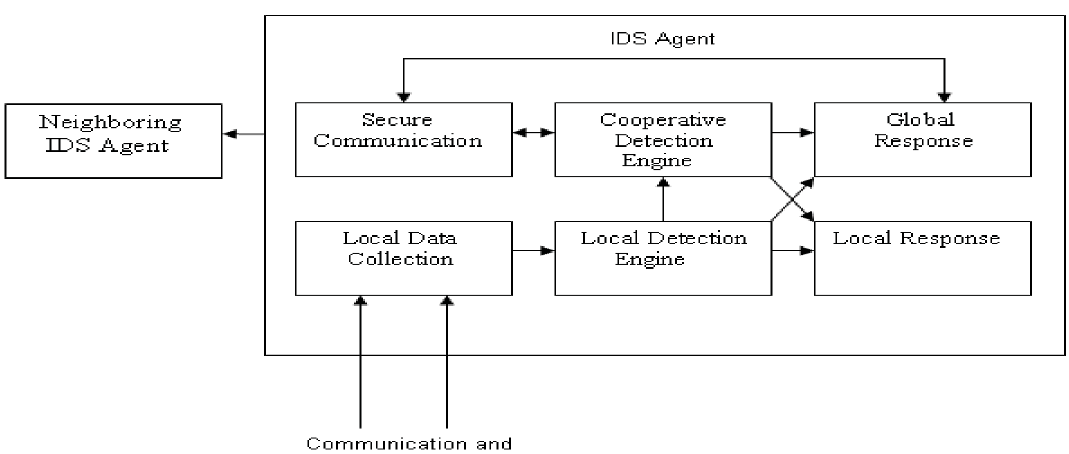

Figure 3 : Cooperative and Distributed Model of an Intrusion Detection System.

agent can make request to the neighboring IDS agents for cooperation through a module named as a cooperative detection engine. This module will help for communicating with other neighboring agents through another module named as a secure communication module.

In 2007 R. Puttini proposed a fully Distributed Intrusion Detection System (DIDS) for mobile adhoc network[20].In this attack detection system distribution is not only on the basis of data collection but there is also execution of the detection algorithm as well as alert correlation. Each node in mobile adhoc network runs a local intrusion detection system (LIDS). All the local intrusion detection systems work with each other in cooperative manner. A mobile agent is used to compensate with the dynamic state of high mobility nodes in wireless network. In this distributed IDS R. Puttini used three types of attacks to show the IDS mechanisms. Intrusion detection is described with the help of data collection, number of attack signatures associated with this data, correlation and alert generation.

In 2010 R.Nakkeeran proposed a new model named as "Agent Based cooperative and distributive model"[16]. In this model three techniques are provided for security solution to neighboring node, current node and global network. The different modules are explained in following section.

i. *Home Agent*

This agent is part of each system and helpful in gathering information about its system which is from application layer to routing layer.

a. *Current Node*

The purpose of Home Agent in each system to monitors its system continuously. If an intrusion or attacker sends some packets to get information or try to broadcast through the system then home agent will call the classifier for finding the intrusions in the network. If there is an attack then it will filter the required system from the global network.

b. *Neighboring Node*

In a network any system can transfer the information to another system through intermediate System. Before transferring the information it send mobile agents to neighboring node for gathering information for finding out the attacks or intrusions. If there is no any intrusion in the system then it will transfer or broadcast the message to neighboring node.

c. *Data collection*

This module is used in each anomaly detection subsystem for collecting values for corresponding layer in the system. Based on the data collected during the normal scenario normal profile is created and during the attack scenario attack data is collected.

d. *Data preprocess*

The audit data is collected in some file and it is used for intrusion detection. In Data preprocess module information is processed with the test data. This preprocessing technique is used for entire layer intrusion detection systems.

ii. *Cross feature analysis for classifier sub model construction.*

iii. *Local Integration*

This module concentrate only on self system and it is responsible for finding local intrusions only. In wireless network each system follows the same method to provide secure global network.

iv. *Global integration*

This module is used for finding out the attacks for entire network. The objective of global integration is to use the results of neighboring nodes for taking decision .The results are used by response module to provide response.

Jelena Mirkovic et al. [21] have proposed a distributed system for DDoS defense, called DefCom. DefCOM nodes spam source, victim and core networks and cooperate via an overlay to detect and stop attacks. Attack response was twofold: defense nodes constrain the attack traffic, relieving victim's resources; they have also cooperated to detect legitimate traffic within the suspicious stream and ensure its correct delivery to the victim. DefCOM design has a solid economic model where networks deploying defense nodes directly benefit from their operation. DefCOM further offers a framework for existing security systems to join the overlay and cooperate in the defense. These features have created a execellent motivation for wide deployment, and the possibility of large impact on DDoS threat.

University of California, U.S. Air Force and Lawrence Livermore Laboratory jointly proposed Distributed Intrusion Detection System (DIDS)[22].DIDS incorporates a monitor on LAN, a monitor on each host and a DIDS director. Host monitor consist of two parts one is host agent and another is host event generator. The purpose of host event generator is to review the audit data from host. This audit data is used for indication of events which are responsible foe attack. This information is reported to DIDS director by Host Agent. LAN monitor consists of LAN agent and LAN event generator. LAN event generator is unlike with the host event generator. It monitors all network traffic, which include host to host connections and different resources used. LAN agent sends the information generated by LAN event generator to the DIDS director. The DIDS director is the heart of Distributed intrusion detection system. DIDS Director further consists of three components that are communication manager, user interface and an expert system. The purpose of the

Different Distributed Intrusion Detection Systems	References of DIDS
Effective Intrusion Detection System using Multiple Sensors in Wireless Network	IDS1
a Cooperative and distributed intrusion detection system for wireless networks.	IDS2
a fully Distributed Intrusion Detection System (DIDS) for mobile adhoc network	IDS3
Agent Based cooperative and distributive Distributed Intrusion Detection System	IDS4
Distributed Intrusion Detection System for detecting denial of service (DDoS) attacks.	IDS5
Distributed Intrusion Detection System (DIDS) used for wireless LAN.	IDS6

Table 1 : References of different Distributed Intrusion Detection Systems.

communication manager is to collect the information from LAN monitors and the host monitors. After collecting the information Communication manager forward this to Expert system. Expert systems do analysis of this information. The expert system in DIDS is a rule based system whose purpose is to analyze the information received from monitors and report to security officials. The user interface allows receiving different reports from expert system, a security official to review the status and can also request additional information related to security of the system. One of the main elements of DIDS is Network User Identification (NID). NID is used to establish an identifier for all users to when they are initially logged in the network. This is used because many attackers use different accounts for making attack in a network. Once a user is logged in to a network, at the same time a NID is assigned to it. Different activities of that user are attributable through NID. If user logged in again by another name then its activities can be compared. NID has the potential to track any intruder through no. of hosts.

VII. Comparison of Different Distributed Intrusion Detection Systems

There are a lot of advantages and disadvantages of different distributed intrusion detection systems. Different distributed intrusion detection systems and there references are shown in table 1 and comparison of these systems is shown in table 2.

VIII. Conclusion and Future Work

Only intrusion detection and prevention techniques are not sufficient for securing wireless network but there is also need of good Intrusion Detection System. From the existing DIDS anomaly based intrusion detection systems are more efficient and economic because of distributed nature of wireless ad hoc network. For better understanding of Distributed Intrusion Detection System

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Reference of IDS	Author	Algorithm	Merits	Demerits	ID Method
IDS1	Kachirski and Guha	Mobile Agent Based.	Better network performance.	Only use anomaly based method.	Anomaly Based
IDS2	Y. Huang	Cluster based Distributed Intrusion Detection scheme.	Improved efficiency in the terms of network overhead and memory usage.	False alarm rates are not mentioned and low performance.	Anomaly Based
IDS3	R.Puttini	A Fully Distributed Algorithm.	Identify the source of packet dropping attack and suitable for MANET.	Very time consuming process to learn program profiles and testing processes.	Signature Based
IDS4	R.Nakkeeran	Agent based Cooperative and Distributive system	Low false alarm rate and performance is better than other IDS.	No description about security issues of mobile agents.	Anomaly Based
IDS5	Jelena Mirkovic	A Distributed System for DDoS Defense.	Ability to detect new attacks and latest misuse signatures.	Faces some challenges like arbitrary definition of abnormal activities.	Signature Based
IDS6	James Cannady and Jay Harrell	Cluster based Intrusion Detection System	Reduces communication overheads and good detection rate.	More complex and ineffective co-ordination between DIDS modules.	Anomaly Based

Table 2 : Comparison Table of different Distributed Intrusion Detection Systems (DIDS).

we have given details of different DIDS. We have also given comparison table of different DIDS according to their performance. Future work will involve developing more intelligent and robust intrusion detection algorithms. We will investigate number of attacks on Intrusion Detection System infrastructure.

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Design, Development and Performance Study of a Microcontroller-Based Home Security System Using Mobile Phone

By Tabinder Akter, Mahfuja Akter, Mohammad Mozammel Hoque, Md. Afzalur Rab, & Dr.Md.Habibur Rahman

University of Dhaka

Abstract - Security has been an important issue in the smart phone applications. Here, a security system has been developed that uses a sensor to detect a security violation and informs the owner by calling to his mobile. The system consists of a sensor, microcontroller IC, some relays and a mobile phone that operates when the sensor sense some unwanted breaking of doors, lockers etc. As soon as this occurs the system automatically calls the owner through the mobile phone used in the security system. The system has been designed and developed using locally available components and its performance has been studied. A program has been developed using Flow Code software to drive the system.

Keywords : MSS, Sensor, Relay, PIC microcontroller, Flow code.

GJRE-F Classification : FOR Code: 090601,090699

Design, Development and Performance Study of a Microcontroller-Based Home Security System Using Mobile Phone

Tabinder Akter [α], Mahfuja Akter [σ], Mohammad Mozammel Hoque [ρ], Md. Afzalur Rab [ω], Dr.Md.Habibur Rahman [¥]

Abstract - Security has been an important issue in the smart phone applications. Here, a security system has been developed that uses a sensor to detect a security violation and informs the owner by calling to his mobile. The system consists of a sensor, microcontroller IC, some relays and a mobile phone that operates when the sensor sense some unwanted breaking of doors, lockers etc. As soon as this occurs the system automatically calls the owner through the mobile phone used in the security system. The system has been designed and developed using locally available components and its performance has been studied. A program has been developed using Flow Code software to drive the system.

General Terms : Microcontroller based Security System (MSS), Home security system (HSS), Mobile keypad switch layout and Peripheral interface controller (PIC).

Keywords : MSS, Sensor, Relay, PIC microcontroller, Flow code.

I. Introduction

Homes without security systems are more likely to be targeted by a burglar and most of residential burglaries occur during the daylight. Under smart and intelligent home environment [1], numerous sensors, motion detectors, smoke detectors, water leakage detectors etc., and communication devices can be utilized for connection throughout the house, capable of monitoring and detecting the physical events. The output from these sensors can be used to alert the owner of any unauthorized intrusion or control home appliances such as lightings. Thus, maintaining seamless connectivity between devices and the main controller is very crucial. A lost connectivity can jeopardize the security of the home. It is also an important factor to ensure that, the devices being used operate in very-low power consumption so that they would last longer [2].

In previous years, different home security systems such as Zigbee based security system [3], PIC based security system, SMS based security system [4] etc. has been designed and implemented.

Real time security system based on GSM network [5-7] has also been devised. But the cost of the system is relatively high.

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II. Block Diagram of the System

The block diagram of the system consists of sensor, signal and control unit, interfacing unit and mobile unit.

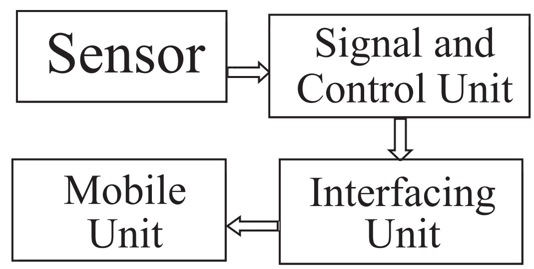

Figure. 1 : Block diagram representation of the Security System

III. System Overview

a) Mobile unit

The numbers of the mobile phone are usually in a row-column array matrix [8]. Numbers 1, 4, 7, 'yes' (a button to dial or receive a call) and * are in the left column. Numbers 2, 5, 8 and 0 are in the middle column and numbers 3, 6, 9 and # are in the right column [9]. The outer circle of the button work as negative terminal and the inner circle of the button work as positive terminal. The negative terminals of all the buttons are shorted to a common. When the positive terminal and the negative terminal are shorted the corresponding numbers are generated. Switching circuit has been used to generate the user number. Copper wires are soldered to keypad buttons which has been connected to relays. For the present research a Motorola mobile phone set have been used [10] which is shown in figure 2 .

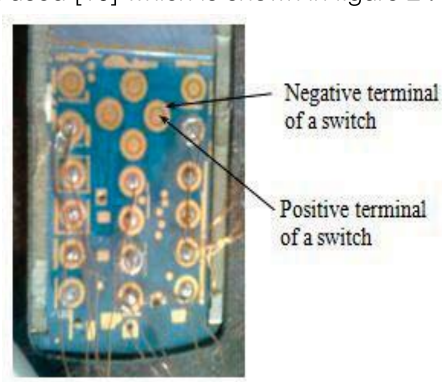

Figure. 2 : Mobile Keypad.

b) Interfacing Unit

As the numbers on mobile phone are in row column array matrix, thus relays [11] (SPDT type) have been used to generate the called number. Each common point of relay is connected to the negative terminal of the button and normally open point of relay is connected to the positive terminal of the button. For example, we want to generate 1, so we have connected negative of the 'yes' button (as negative terminals of same column are common) to the common point of relay and positive of 1 button to the normally open point of relay [11]. Like this way we have connected all the other digits to the relays.

Relay's coils are connected to the collectors of C828 transistors. Transistor has been designed as a switch. A diode is connected to the collector for protection of the transistor. Emitter is grounded and two resistances of 10k are connected to the base of transistor as shown in figure 3. The controlling signal generated by a microcontroller and is aplied to register R_1.

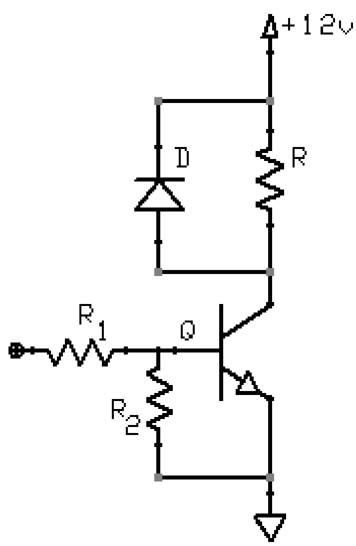

Figure. 3 : Mobile Keypad

c) The Signal and Control Unit

The control system can sense the opening of a door. In this system a very thin wire has been used to detect the condition of the door. The wire will be attached with the door and when the door is opened the wire will be disconnected from the circuit and will be sensed. One end of the pin RA0 (pin 2) of the microcontroller and the other end to ground [12]. A 1kΩ resistor is connected from pin RA0 to 5 volt. Hence, when the wire is connected in the circuit the logic level of pin2 will be zero and when the wire is detached the logic level of pin2 will be 1. Microcontroller always checks the logic levels of this pin [13].

When the pin is at zero level, the controller waits a few seconds and checks. The loop will be continued until the wire is detached and that the door has been opened. When the door is opened the wire will be detached and the microcontroller will find a logic 1 at pin RA0 and it will start to call a particular (set by user) mobile phone number.

In this work, PORT-A of the microcontroller has been chosen as an input port and PORT-B and PORT-C has been chosen as output port [14].

The transistor switch has been connected to the output of the microcontroller. Microcontroller is operated at 5 volt whereas the relays work with 12V. For this reason we have used a regulator IC to generate 5 volt from 12 volt. The 7805 IC was used as a regulator IC.

To generate common numbers we have used diode at the output of the microcontroller. A switch at pin 1 of microcontroller has also been used for refreshment purpose.

IV. Working Diagram

Fig. 4 shows the complete circuit diagram where mobile phone section has not been added.

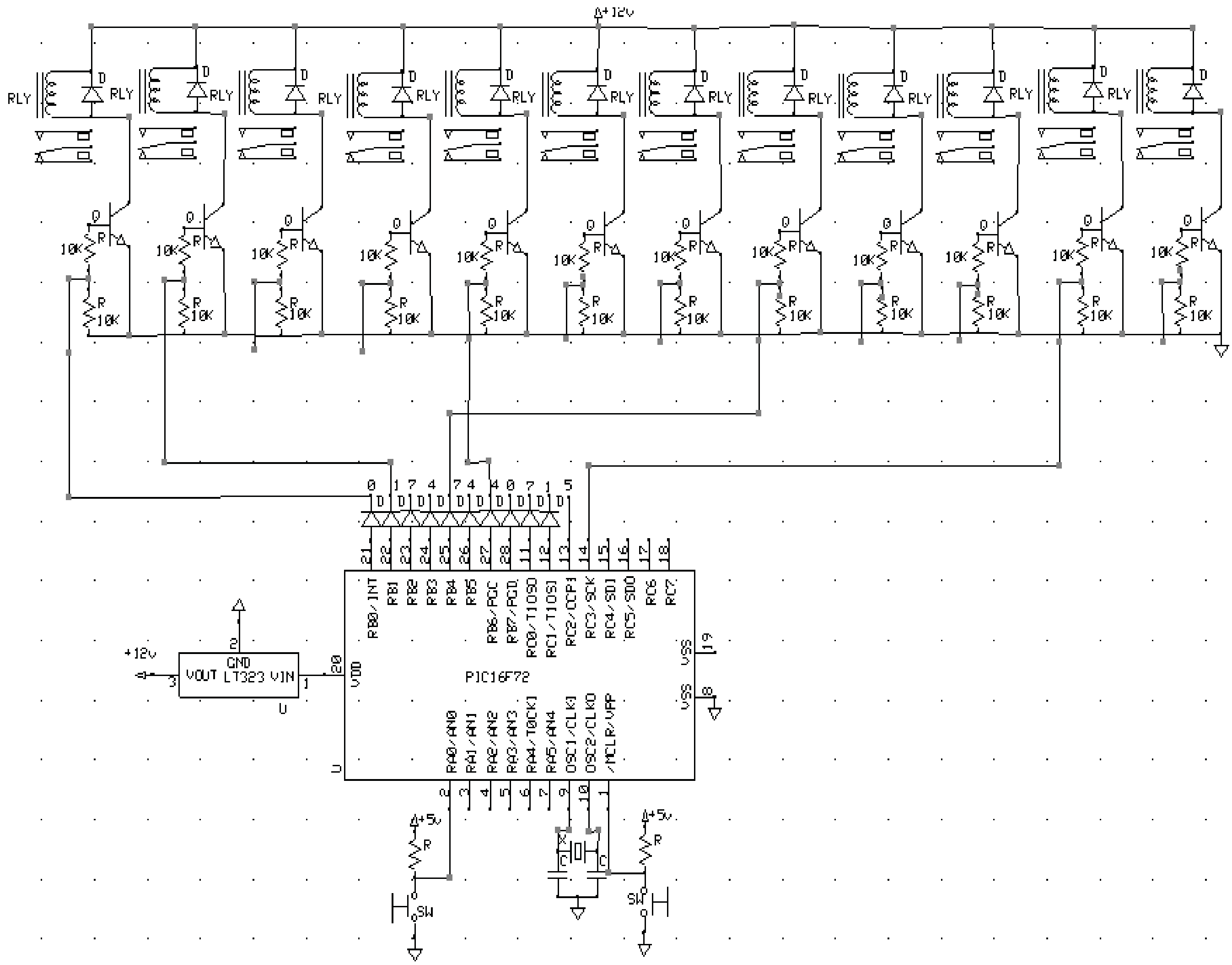

Figure. 4 : working diagram of MSS.

V. Operation of the Whole System

In the present design, a wire has been used as a sensor which is connected to the main circuit. When the wire is cut, according to the program the output will be activated one after another.

According to the output of microcontroller the transistor switch is then activated and mobile button is pressed. Thus a call has been made.

To generate each number one second is required. That means, to generate eleven numbers eleven seconds is required. Microcontroller IC has been programmed for this purpose. Here we use yes button to generate the call. For this purpose we use 12 relays which are connected to the mobile phone keyboard pad.

In case when it is necessary to generate the mobile number 01749757937, then the relay contacts are connected to keys 0, 1, 7, 4, 9, 7, 5, 7, 9, 3 and 7.

When the yes button is pressed, the number will be started to generate. After generating the number, we have wait for few seconds for generating the call. It is a continuous process.

That means the call is repeatedly generated as long as the user will not disconnect it.

a) Timing calculation

For generating each number, one second is required. So the time to generate 11 numbers is 11 seconds. Time taken to press the 'yes' button is 1 second. So, time taken to dial a number is 12 seconds.

Considering 8 seconds as time required to generate a call, about 20 seconds is required to ring the owners mobile. Thus it performs quite well.

VI. Cost Analysis

For the proposed system, components of the system are available in local market. The recent cost for entire the system is given below.

Table.1 : Production cost of MSS system.

Name of the Components	Type/ Model No.	Quantity	Total Prices in TK.
Microcontroller	PIC16F72	1	100
Crystal Oscillator	4.00 MHz	1	20
Switches	Mega Size	2	40
Transistor	C828	12	60
Resistors	470Ω, 1KΩ	21,6	10
Decoder IC	SN74LS47	3	60
Counter IC	DM74LS90	2	40
Power Supply	5V	1	45
PCB	------------		20
Connecting Wire	------------	-- -----	20
7 Segment Display	Common Anode	3	45
Others	-------------	--------	50
			Total amount = 475

The above data clearly shows the cost superiority of import system to locally developed system.

b) Low Cost

The MSS (Microcontroller based Security System) is a security system for homes, offices, shops, banks etc. It is developed to make offices and especially homes much more secure. Although there are existing security systems for that place, the MSS differ from them in many ways. The system was designed using locally available components and it is very cheap. Accessibility from mobile devices makes the MSS is really different from existing security system. The MSS has lots of beneficial effect on society. Its social impact will be very important, because people far away from their home need not to be worried about it. In the time of emergency they will be warned by the system.

It is a law cost security system, and it really easy to make a home secured with MSS.

c) Future work

The system was an experimental platform; which have successfully implemented and tested all the main functions that the system is intended to meet. In its commercial release, the system may lead to great achievements in home and office security and prevention of different dangerous situations such as fire and theft.

In further release of the HSS, the system can be extended to transmit camera view of home to the users via their mobile phones and other mobile devices like PDA [15].

The system may consist of a gas/smoke sensor, an aqua sensor, a temperature sensor, a microcontroller module, a number of cameras, a GPRS modem and a PC. Sensor can be wired to the microcontroller module, and this module can be connected to PC via its serial port. These sensors can cause corresponding actions to be taken by the main software on PC. The cameras placed rooms by the users, connected to PC from USB port. GPRS modem, used by software to send SMS to users and call police and fire departments, can be connected to the PC via its serial port (COM1). The microcontroller, cameras and GPRS modem can be connected to this PC. Figure-5 shows the components and structure of the future work of a Security System.

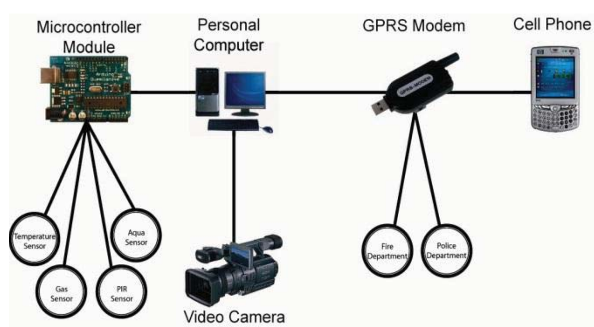

Figure-5 : Future work

VII. Conclusion

The MSS (Microcontroller based Security System) is a security system for homes, offices, shops, banks etc [10]. It is developed to make offices and especially homes much more secure. Although there are existing security systems for that place, the MSS differ from them in many ways. The system was designed using locally available components and it is very cheap. Accessibility from mobile devices makes the MSS is really different from existing security systems. The MSS has lots of beneficial effect on society. Its social impact will be very important, because people far away from their home need not to be worried about it. In the time of emergency they will be warned by the system.

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Isolated bidirectional full-bridge dc–dc converter with fly back snubber for high-power applications

By K.Banu priya, JBV Subrahmanyam, Ch.Srikanth & PK Sahoo Bandana

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Abstract - This paper introduces a flyback snubber to recycle the absorbed energy in the clamping capacitor. The flyback snubber can be operated independently to regulate the voltage of the clamping capacitor; therefore, it can clamp the voltage to a desired level just slightly higher than the voltage across the low-side transformer winding. Since the current does not circulate through the full-bridge switches, their current stresses can be reduced dramatically under heavy-load condition, thus improving system reliability significantly.

Keywords : flyback snubber, clamping capacitor, transformer winding, full-bridge switches, system reliability.

GJRE-F Classification : FOR Code: 090604,090699

ISOLATEDBIDIRECTIONALFULL-BRIDGEDCDCCONVERTERWITHFLYBACKSNUBBERFORHIGH-POWERAPPLICATIONS

Strictly as per the compliance and regulations of:

Isolated bidirectional full-bridge dc–dc converter with fly back snubber for high-power applications

K.Banu priya [α], JBV Subrahmanyam [σ], Ch.Srikanth [ρ] PK Sahoo [Ѡ] & Bandana [¥]

Abstract - This paper introduces a flyback snubber to recycle the absorbed energy in the clamping capacitor. The flyback snubber can be operated independently to regulate the voltage of the clamping capacitor; therefore, it can clamp the voltage to a desired level just slightly higher than the voltage across the low-side transformer winding. Since the current does not circulate through the full-bridge switches, their current stresses can be reduced dramatically under heavy-load condition, thus improving system reliability significantly.

Keywords : flyback snubber, clamping capacitor, transformer winding, full-bridge switches, system reliability.

I. Introduction

Power electronic converters are used extensively in personal electronics, power systems, hybrid electric vehicles (HEVs), and many other applications to provide dc voltage sources and manage power flow by switching actions .To obtain high power quality, switching control strategies that can achieve high performances are attracting more and more attention .[1]

Many advanced control strategies, such as fuzzy-neural control or sliding-mode control, have been proposed to enhance the steady-state and dynamic performance of power electronic systems. Although these control strategies are predicted to be promising in more complex-structured converters, such as dual-active-bridge (DAB) and dc–dc converters. Most of the present applications are still confined to simple structured circuits, such as buck, boost, and half-bridge converters.[1]

Compared to traditional dc–dc converter circuits, isolated bidirectional DAB dc–dc converters have many advantages, such as electrical isolation, high reliability, ease of realizing soft-switching control, and bidirectional energy flow.[1]

A double-phase-shift control for a unidirectional three-level converter is proposed in. The phase shift is implemented on the primary side. A start-up circuit to suppress the inrush current with a set of auxiliary circuits is proposed.[2]

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The dc–dc converter is a key component in hybrid electric vehicles (HEVs) to manage power flow and maintain battery health. Electrical isolation may be required to provide safe operation for the equipment operated on the hybrid battery, such as in military applications. State-of-the-art isolated dc–dc converters are generally based on single-phase full-bridge topologies with isolation transformers. An isolated bidirectional dc–dc converter, which consists of dual H-bridges located on the primary and secondary sides of an isolated transformer, respectively.[3]

The primary bridge consists of four switches, Q1, Q2, Q3 and Q4, which are commonly insulated gate bipolar transistors (IGBTs) for high-power applications. The second H-bridge also consists of four switches, Q5, Q6, Q7, and Q8, which are connected to the secondary winding of the transformer. With a phase-shift control algorithm, the first H-bridge provides a square wave with duty ratio of 50% to the primary winding of the high-frequency transformer.[3]

In traditional unidirectional dc–dc converters, the power ratings are generally low, and the switching frequency is relatively high (for MOSFET or Si C, turning on and off processes are both in the nanosecond level). Therefore, there is, generally, no need to deal with dead band effect. However, in high-voltage and high-power isolated bidirectional dc–dc converters, the dead band and phase-shift error will greatly affect the operation of the converter, both in steady-state and transient processes. These issues generally deteriorate the operational performance, or even damage the system under some specific switching conditions because of large unexpected current and voltage spikes.[4]

A few integrated multi-port dc-dc converter topologies are found in the literature. There are two categories for the integrated isolated multi-port converter. One type of converter involves a transformer in which there is a separate winding for each port, therefore all ports are fully electrically isolated. The other type has a reduced parts count where some windings are absent, if the system allows the corresponding ports to share a common ground.[4]

A dual active full bridge dc-dc converter was proposed for high power BDC , which employs two voltage-fed inverters to drive each sides of a transformer. Its symmetric structure enables the

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bidirectional power flow and ZVS for all switches. A dual active half bridge current-voltage-fed soft-switching bidirectional dc-dc converter was proposed with reduced power components however, the current-fed half bridge suffers from a high voltage spike because of the leakage inductance of the transformer. When the voltage amplitude of the two sides of the transformer is not matched, the current stresses and circulating conduction losses become higher.[5]

In addition, these converters cannot achieve ZVS in low-load condition. These disadvantages make it not suitable for large variation of input or output voltage condition. An asymmetry bidirectional dc-dc converter with Phase shift plus PWM (PSP) control was proposed in, the circulating conduction loss is reduced. The converter with an active clamping branch avoids the voltage spike, achieves Zero Voltage Switching and restrains the start-inrush current.[6]

The demands of a bidirectional dc/dc converter are high frequency, high power density, high efficiency and high reliability. Nevertheless, the conventional bidirectional dc/dc converters still have some drawbacks: Electric insulation and soft switching is difficult to realize, and the reverse-recovery effect of the rectifier diode restricts the switching speed. These defects limit the high-frequency power conversion applied in a bidirectional dc/dc converter. Therefore, an isolated bidirectional dc/dc converter with soft switching is the best way to meet the previously mentioned demands.[7]

II. Configuration & Operation

The proposed isolated bidirectional full-bridge dc–dc converter with a fly back snubber is shown in Fig.1 The converter is operated in two modes: buck mode and boost mode. Fig.1 consists of a current-fed switch bridge, a fly back snubber at the low-voltage side, and a voltage-fed bridge at the high-voltage side. Inductor *Lm* performs output filtering when power flows from the high-voltage side to the batteries, which is denoted as a buck mode. On the other hand, it works in boost mode when power is transferred from the batteries to the high-voltage side. Furthermore, clamp branch capacitor *CC* and diode *DC* are used to absorb the current difference between current-fed inductor *Lm* and leakage inductance *Lll* and *Llh* of isolation transformer *Tx* during switching commutation. The fly back snubber can be independently controlled to regulate *VC* to the desired value, which is just slightly higher than V_{AB}. Thus, the voltage stress of switches *M*1–*M*4 can be limited to a low level. The major merits of the proposed converter configuration include no spike current circulating through the power switches and clamping the voltage across switches *M*1–*M*4, improving system reliability significantly. Note that high spike current can result in charge migration, over current density, and extra magnetic force, which will deteriorate in MOSFET carrier density, channel width, and wire bonding and, in turn, increase its conduction resistance. A bidirectional dc–dc converter has two types of conversions: step-up conversion (boost mode) and step-down conversion (buck mode). In boost mode, switches *M*1–*M*4 are controlled, and the body diodes of switches *M*5–*M*8 are used as a rectifier. In buck mode, switches *M*5–*M*8 are controlled, and the body diodes of switches *M*1–*M*4 operate as a rectifier. To simplify the steady-state analysis, several assumptions are made, which are as follows.

1. All components are ideal. The transformer is treated as an ideal transformer associated with leakage inductance.
2. Inductor *Lm* is large enough to keep current *iL* constant over a switching period.
3. Clamping capacitor *CC* is much larger than parasitic capacitance of switches *M*1–*M*8 [7]

III. Step-up Conversion

In boost mode, switches *M*1–*M*4 are operated like a boost converter, where switch pairs (*M*1 , *M*2) and (*M*3 , *M*4) are turned ON to store energy in *Lm*. At the high-voltage side, the body diodes of switches *M*5–*M*8 will conduct to transfer power to *VHV* . When switch pair (*M*1 , *M*2) or (*M*3 , *M*4) is switched to (*M*1 , *M*4) or (*M*2 , *M*3), the current difference *iC* (= *iL* − *ip*) will charge capacitor *CC* , and then, raise *ip* up to *iL* . The clamp branch is mainly used to limit the transient voltage imposed on the current-fed side switches. Moreover, the fly back converter can be controlled to charge the high-voltage-side capacitor to avoid over current. The clamp branch and the fly back snubber are activated during both start-up and regular boost operation modes. A non phase-shift PWM is used to control the circuit to achieve smooth transition from start-up to regular boost operation mode. Referring to Fig, the average power *PC* transferred to *CC* can be determined as follows:

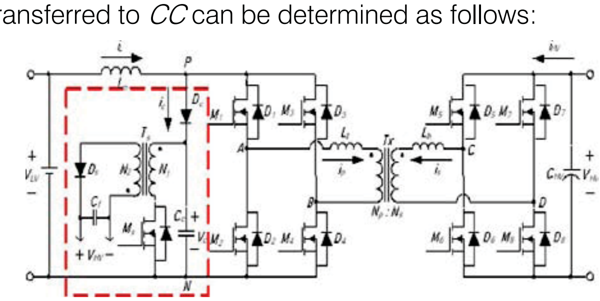

Fig.1 : Isolated bidirectional full-bridge dc–dc converter with a fly back snubber

$$P_C = \frac{1}{2} C_C \left[(i_L Z_o)^2 + 2 i_L Z_o V_{C(R)} \right] f_s \qquad (1)$$

where

$$Z_o = \sqrt{\frac{L_{eq}}{C_C}}$$

$$L_{eq} = L_{ll} + L_{lh} \frac{N_P^2}{N_S^2}$$

$V_{C(R)}$ stands for a regulated V_C voltage, which is close to (V_{HV} (N_P /N_S)), f_s is the switching frequency, and L_m_L_{eq}. Power P_C will be transferred to the high-side voltage source through the fly back snubber, and the snubber will regulate clamping capacitor voltage V_C to V_C (R) within one switching cycle T_s (=1/f_s). Note that the fly back snubber does not operate over the interval of inductance current i_p increasing toward i_L. The processed power P_C by the fly back snubber is typically around 5% of the full-load power for low-voltage applications. With the fly back snubber, the energy absorbed in C_C will not flow through switches M1–M4, which can reduce their current stress dramatically when L_{eq} is significant. Theoretically, it can reduce the current stress from 2i_L to i_L . The peak voltage V_C (P) of V_C will impose on M1–M4 and it can be determined as follows:

$$V_{C(P)} = i_{L(M)} Z_o + V_{HV} \frac{N_p}{N_s} \tag{2}$$

Where i_L (M) is the maximum inductor current of i_L, which is related to the maximum load condition. Additionally, for reducing conduction loss, the high-side switches M5–M8 are operated with synchronous switching. Reliable operation and high efficiency of the proposed converter are verified on a prototype designed for alternative energy applications. The operation waveforms of step-up conversion are shown in Fig.6 A detailed description of a half-switching cycle operation is shown in fig.2

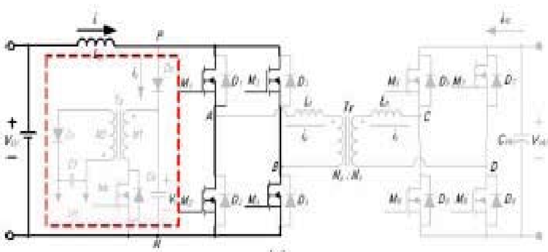

Fig.2 : Boost modes 1and 2

Mode 1 [t0≤ t < t1]: Mode 2 [t1≤ t < t2]:

In these modes, all of the four switches M1–M4 are turned ON. Inductor Lm is charged by VLV, inductor current iL increases linearly at a slope of VLV /Lm, and the primary winding of the transformer is short-circuited. The equivalent circuit is shown in Fig.2

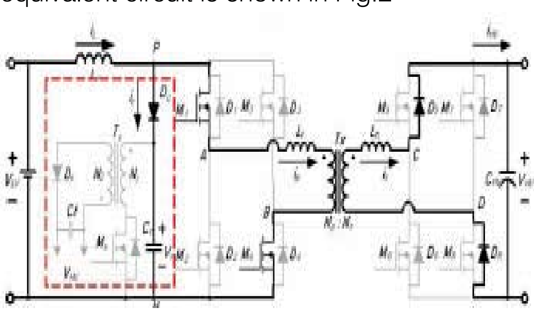

Fig.3 : Boost mode 3 [t2<t<t3]

Mode 3 [t2 ≤ t < t3]:

At t2, clamping diode Dc stops conducting, and the fly back snubber starts to operate. At this time, clamping capacitor Cc is discharging, and fly back inductor is storing energy. Switches M1 and M4 still stay in the ON state, while M2 and M3 remain OFF. The body diodes of switch pair (M5 , M8) remain ON to transfer power. The equivalent circuit is shown in Fig.3

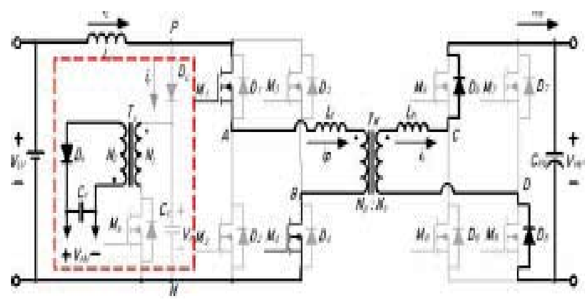

FIG 4 : BOOST MODE 4 [t3<t<t4]

Mode 4 [t3 ≤t<t4]:

At t3, the energy stored in fly back inductor is transferred to the high-voltage side. Over this interval, the fly back snubber will operate independently to regulate VC to VC (R). On the other hand, switches M1 and M4 and diodes D5and D8 are still conducting to transfer power from VLV to VHV .The equivalent circuit is shown in Fig.4

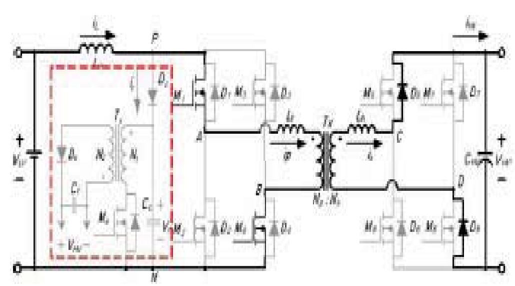

FIG 5 : BOOST MODE 5 [t4<t<t5]

Mode 5 [t4 ≤ t < t5]:

At t4, capacitor voltage VC has been regulated to VC (R) , and the snubber is idle. Over this interval, the main power stage is still transferring power from VLV to VHV. It stops at t5 and completes a half-switching cycle operation.[7]The equivalent circuit is shown in Fig.5

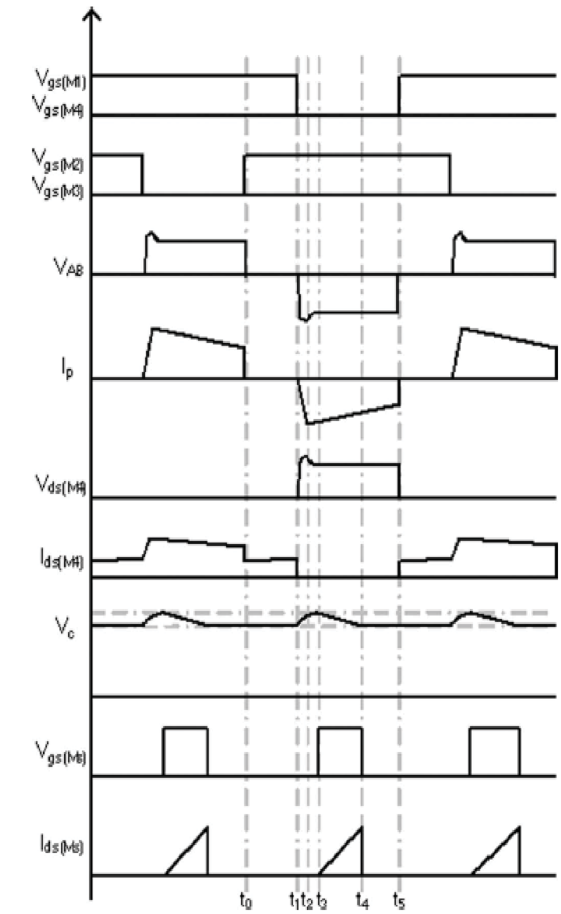

Fig.6 : Operation waveforms of step-up conversion.

IV. Step-Down Conversion

In the analysis, leakage inductance of the transformer at the low-voltage side is reflected to the high-voltage side, as shown in Fig. 4, in which equivalent inductance L_{eq} equals(L_{lh} + L_{ll} ($N2p\_N2\ s$)). This circuit is known as a phase-shift full-bridge converter. In the step-down conversion, switches $M5$–$M8$ are operated like a buck converter, in which switch pairs ($M5$, $M8$) and ($M6$, $M7$) are alternately turned ON to transfer power from VHV to VLV. Switches $M1$–$M4$ are operated with synchronous switching to reduce conduction loss. For alleviating leakage inductance effect on voltage spike, switches $M5$–$M8$ are operated with phase-shift manner. Although, there is no need to absorb the current difference between iL and ip , capacitor CC can help to clamp the voltage ringing due to L_{eq} equals (L_{ll} + L_{lh} ($N2p\_N2s$)) and parasitic capacitance of $M1$–$M4$ .The operation waveforms of step-down conversion are shown in Fig.12. A detailed description of a half-switching cycle operation is shown as follows.

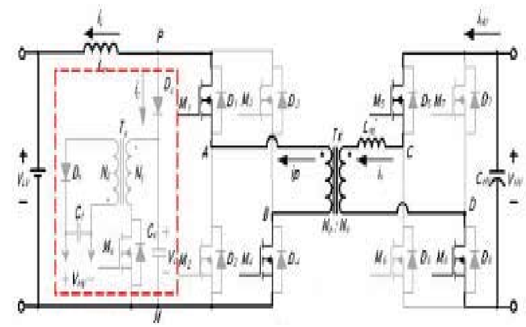

Fig.7 : Buck mode 1

Mode 1 [t0 ≤ t < t1]:

In this mode, M5 and M8 are turned ON, while M6 and M7 are in the OFF state. The high-side voltage VHV is immediately exerted on the transformer, and the whole voltage, in fact, is exerted on the equivalent inductance L*eq and causes the current to rise with the slope of VHV /L*eq. With the transformer current increasing linearly toward the load current level at t1 , the switch pair (M1 , M4) are conducting to transfer power, and the voltage across the transformer terminals on the current-fed side changes immediately to reflect the voltage from the voltage-fed side, i.e., (VHV (Np /Ns)). The equivalent circuit is shown in Fig.7

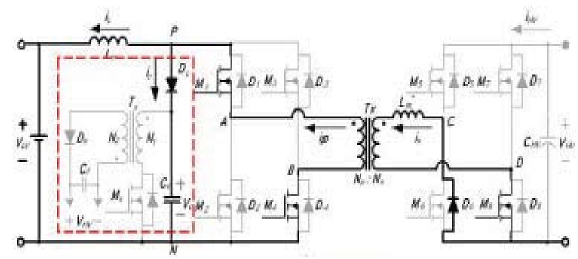

Fig.8 : Buck mode 2

Mode 2 [t1 ≤t<t2]:

At t1,M8 remains conducting, whileM5 is turned OFF. The body diode of M6 then starts to conduct the freewheeling leakage current. The transformer current reaches the load-current level at t1 , and VAB rise to the reflected voltage (VHV (Np /Ns)). Clamping diode Dc starts to conduct the resonant current of Leq and the clamp capacitor CC . This process ends at t2 when the resonance goes through a half resonant cycle and is blocked by the clamping diode Dc. The equivalent circuit is shown in Fig.8

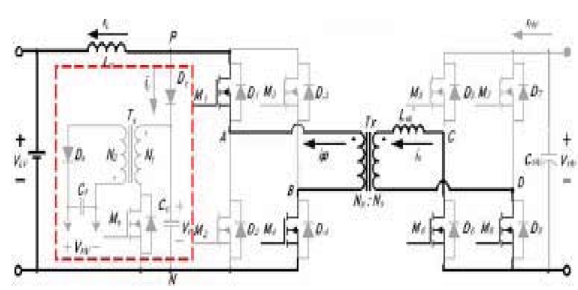

FIG. 9 : Buck Mode 3

Mode 3 [t2 ≤ t < t3]:

At t2 , with the body diode of switchM6 conducting, M6 can be turned ON with zero-voltage switching (ZVS). The equivalent circuit is shown in Fig.9

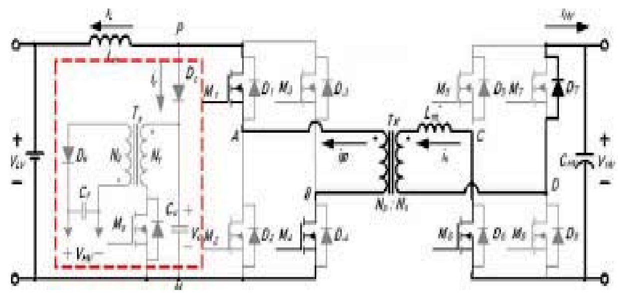

FIG.10 : Buck Mode 4

Mode 4 [t3 ≤t<t4]:

At t3 ,M6 remains conducting, whileM8 is turned OFF. The body diode of M7 then starts to conduct the freewheeling leakage current. The equivalent circuit is shown in Fig.10

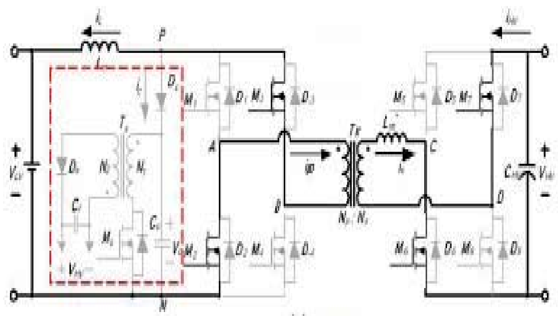

Fig.11 : Buck Mode 5

Mode 5 [t4 ≤ t < t5]:

At t4 , with the body diode of switch M7 conducting, M7 can be turned ON with ZVS. Over this interval, the active switches change to the other pair of diagonal switches, and the voltage on the transformer reverses its polarity to balance flux. It stops at t5 and completes a half-switching cycle operation. [7]
The equivalent circuit is shown in Fig.11

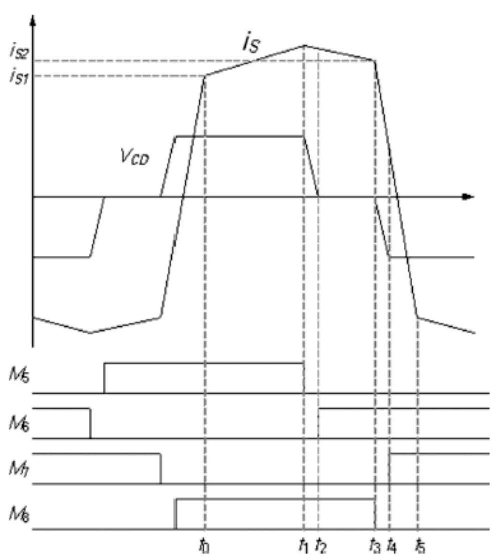

Fig. 12 : Operation waveforms of step-down conversion

a) Practical Consideration

i. *Low-Voltage Side*

Switch pairs (M1,M4) and (M2,M3) are turned ON alternately under any load condition. Its minimum conduction time is

$$T_{C(\min)} = \frac{L_{eq} i_L}{V_{AB}}. \tag{3}$$

ii. *Clamping Capacitor*

For absorbing the energy stored in the leakage inductance and to limit the capacitor voltage to a specified minimal value Vc, l, capacitance Cc has to satisfy the following inequality

$$C_c \geq \frac{L_{eq}(i_L - i_P)^2}{V_{C,l}^2}. \tag{4}$$

iii. *Fly back Converter*

In the interval of t1 ≤ t ≤ t2 , the high transient voltage occurs inevitably in boost mode, which could be suppressed by the clamp branch (Dc , Cc). The energy stored in capacitor Cc is transferred to the high-voltage side via a fly back converter. The regulated voltage level of the fly back converter is set between 110%–120% of the steady-state voltage at the low-voltage side. Power rating of the fly back converter can be expressed as follows:

$$P_{FB} = 0.5 C_c (V_{c,h}^2 - V_{c,l}^2) f_s \tag{5}$$

where Vc,h is the maximum voltage of Vc , Vc,l is the minimum voltage of Vc , and fs is the switching frequency.[7]

V. Simulation Results

Fig.17 : Illustrates the simulated circuit

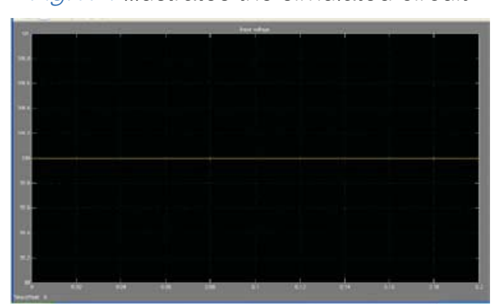

Fig.13 : Input voltage

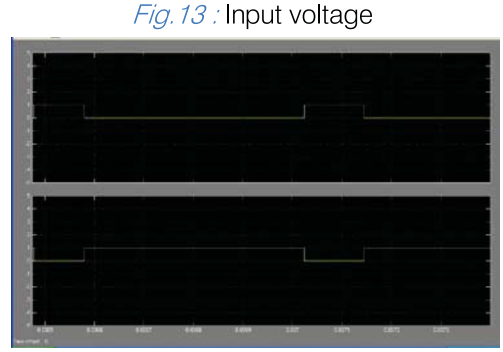

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Fig.14 : Pulse waveform

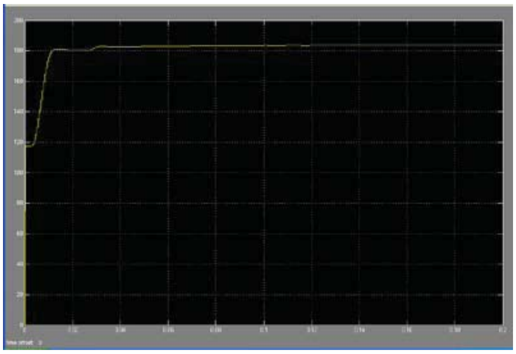

Fig.15 : Capacitor voltage

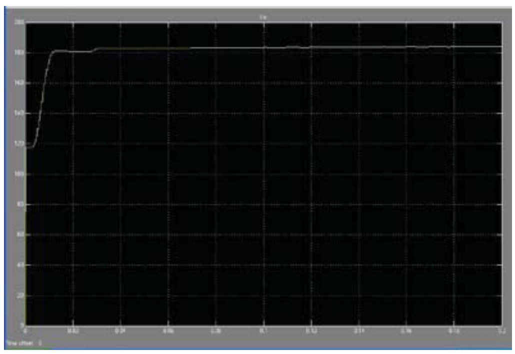

Fig.16 : Output voltage

Fig.13 to 16 confirm the applicability of the proposed method

VI. Conclusion

This paper presents an isolated bidirectional full-bridge dc–dc converter with a fly back snubber for high-power applications. The fly back snubber can alleviate the voltage spike caused by the current difference between the current-fed inductor and leakage inductance of the isolation transformer, and can reduce the current flowing through the active switches at the current fed side by 50%. Since the current does not circulate through the full-bridge switches, their current stresses can be reduced dramatically under heavy-load condition, thus improving system reliability significantly. The fly back snubber can also be controlled to achieve a soft start-up feature. It has been successful in suppressing inrush current which is usually found in a boost-mode start-up transition.

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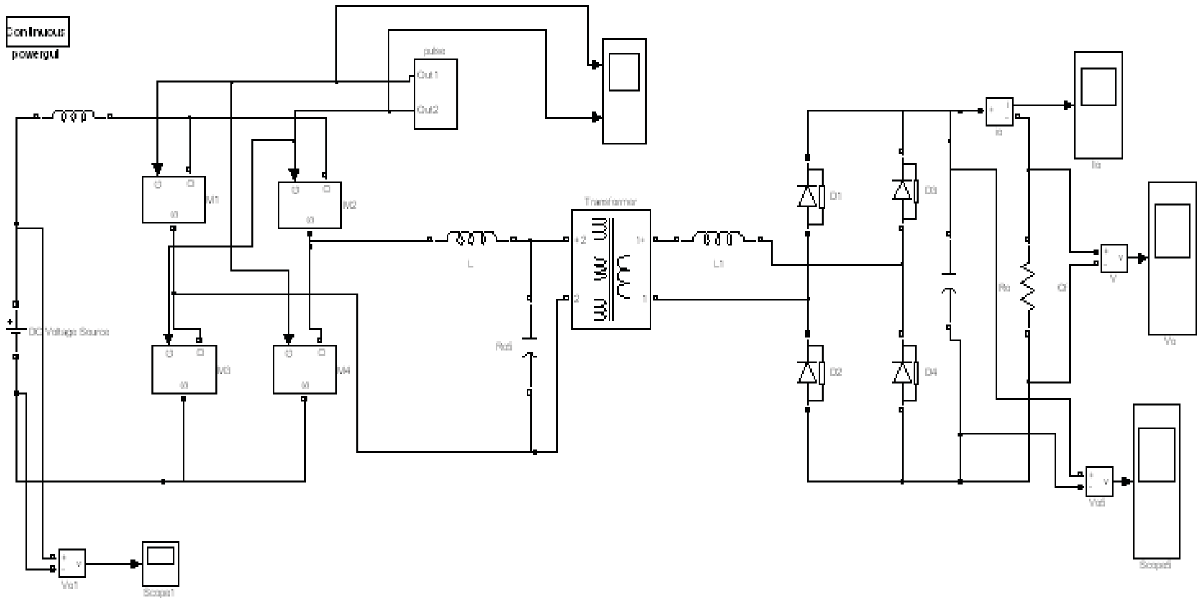

Fig.17 : Simulated circuit

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A Comparative Study of Spread Spectrum Technique based on Various Pseudorandom Codes

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GJRE-F Classification : FOR Code: 100507,100599

A COMPARATIVESTUDYOFSPREADSPECTRUMTECHNIQUEBASEDONVARIOUSPSEUDORANDOMCODES

Strictly as per the compliance and regulations of:

A Comparative Study of Spread Spectrum Technique based on Various Pseudorandom Codes

Vandana Nath [α] & Abhishesh Kumar [σ]

Abstract - **This paper presents a comparative study of Frequency Hopping Spread Spectrum and Direct Sequence Spread Spectrum techniques. The Transmitter and Receiver have been implemented using MATLAB. Maximum-length sequences, Gold Sequences and Walsh Codes have been used as the pseudorandom codes for transmission. Finally received signal have been evaluated on the basis of Bit Error Rate for all the used codes.**

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I. Introduction

Spread Spectrum is a modulation method that spreads a narrowband signal over a wide range of frequencies at the transmitting end and then de-spreads it into the original data bandwidth at the receiving end. The Spread Spectrum technique increases the bandwidth of the transmitting signals to a value much larger than is needed to transmit it. Thus, the power per unit bandwidth is minimized[1]. Spread signals are intentionally made to be much wider band than the information they are carrying to make them more noise-like. As the Spread Spectrum Signal is more noise like, this is the reason that they are harder to detect, intercept or demodulate. It is harder to determine the actual presence of the Spread Spectrum signal[1]. Further, Spread Spectrum signals are harder to jam (interfere with) than narrowband signals. These Low Probability of Intercept (LPI) and anti-jam (AJ) features are why the military has used Spread Spectrum for so many years. Spread spectrum technology allows to bring a number of the same information as that can be transmitted using the narrowband carrier signal and spreaded the signal on a frequency range much greater. Spread Spectrum uses wide band, noise-like sequences called Pseudorandom sequences binary sequences (PRBSs) or pseudo noise (PN) or m-sequences. The PN sequences are semi random sequences i.e. it appears random during its code length and the entire sequence repeats indefinitely. They are generated with the use of Linear Feedback Shift Register (LFSRs)[2].

II. Dsss

In Direct Sequence-Spread Spectrum the baseband signal is combined with a code word,which is known as chipping code. The combined signal is then used to modulate the RF carrier signal. In this way the narrowband signal is spread over the large bandwidth. The original receiver is able to retrieve the correct signal if and only if it knows the correct chipping code (i.e. PN code). The PN is obtained using a PN generator. Frequency of the PN sequence is chosen higher than that of the data signal. This generator consists of a Linear Feedback Shift Registers (LFSRs), and a logic circuit with feedback that determines the PN signal. After spreading, the signal is modulated using a digital modulation technique and then transmitted. BPSK (Binary Phase Shift Keying) is the most widely used modulation scheme in Spread Spectrum Systems. The equation that represents this DS-SS signal using BPSK as modulation is:

$$S(t) = \frac{\sqrt{2E}}{T}\, m(t)p(t)\cos(2\pi f_c t + \theta) \qquad (1)$$

Where $S(t)$ is the transmitted DSSS signal, m(t) is the data sequence, $p(t)$ is the PN spreading sequence, f_c is the carrier frequency, and θ is the carrier phase angle at t=0. Each symbol in m(t) represents a data symbol and has a pulse duration of Ts . Each pulse in $p(t)$ represents a chip sequence, and has duration of Tc. The transitions of the data symbols and chips coincide such that the ratio Ts to Tc is an integer.

III. Fhss

In Frequency Hopped Spread Spectrum, the data is directly modulated onto a carrier frequency. That carrier frequency hops across a number of channels within the RF band using a Pseudo random hopping pattern. That pattern is nothing but the list of different frequencies that is maintained in a table called Hopping table. It is this hopping table by which the carrier jumps from on frequency to another at a specified time interval. The advantage of using this scheme is that the signal sees a different channel and a different set of interfering signals during each hop. This avoids the problem of failing communication at a particular frequency, because of a fade or a particular interferer. The number of frequencies over which the signal may hop is usually a power of 2, although all these frequencies are not necessarily used. The frequency hopping is accomplished by means of a digital frequency synthesizer, which in turn is driven by PN code generator. The M sequence ,i.e. c chips will produce $M=2^c$ frequencies for each distinct combination of these digits.

PN sequence in the receiver should be an exact replica of the one used in the transmitter, with no delays, because this might cause severe errors in the incoming message.

IV. Pseudorandom Pn Sequence

Pseudo random binary sequences ,also known as pseudo noise(PN) or maximal length binary sequences (m sequences) are widely used sequences in the Spread Spectrum systems. In a truly random sequence the bit pattern never repeats, however generation of such a sequence is difficult as it is not practical. The data should appear randomly to the channel and but be predictable to the user. This is therefore the PN sequences are used in the applications. A pseudo random sequence is a semi-random sequence in the sense that it appears random within the sequence length, fulfilling the need of randomness, but the sequence is not actually random. Instead the entire sequence repeats itself indefinitely. To a unintended observer or receiver the actual signal appears as a noise like structure. However the user who is aware of the original PN sequence and its properties will be able to retrieve the correct transmitted signal at its end. The PN sequences are generated by using the Linear Feedback Shift Register (LFSR). An LFSR is most often a shift register whose input bit is driven by the Exclusive OR (XOR) of some bits of the overall shift register value. The initial value of the LFSR is called the seed, and because the operation of the register is deterministic, the stream of values produced by the register is completely determined by its current (or previous) state. Likewise, because the register has a finite number of possible states, it must eventually enter a repeating cycle. However, an LFSR with a well-chosen feedback function can produce a sequence of bits which appears random and which has a very long cycle. If there is 'n' number of flip-flops in the LFSR the code length will be '2^n-1'.

V. Gold Codes

A Gold code, also known as Gold sequence, is a type of binary sequence, have small cross-correlations property within a set, which is useful when multiple devices are broadcasting in the same range [3]. A set of Gold code sequences consists of 2^n-1'. A set of Gold codes can be generated with the following steps. Gold sequences help generate more sequences out of a pair of m-sequences giving now many more different sequences to have multiple users. Pick two maximum length sequences of the same length '2^n-1' such that their absolute cross-correlation is less than or equal to '$2^{(n+2)/2}$' , where 'n' is the size of the LFSR used to generate the maximum length sequence. The set of the '2^n-1' exclusive-ors of the two sequences in their various phases (i.e. translated into all relative positions) is a set of Gold codes. The highest absolute cross-correlation in this set of codes is '$2^{(n+2)/2}+1$' for even 'n' and '$2^{(n+1)/2}+1$'for odd 'n'.

i.e $= 2^{(n+2)/2}+1$ for 'n' is even

$= 2^{(n+1)/2}+1$ for 'n' is odd

The exclusive or of two Gold codes from the same set is another Gold code in some phase. Within a set of Gold codes about half of the codes are balanced - the number of ones and zeros differs by only one.

VI. Walsh Code

Walsh code provides useful code sets for CDMA wireless systems as all the codes of the sets are orthogonal to each other. Walsh functions are generated by mapping codeword rows of special square matrix called Hadamard matrix[4]. The length N of a Walsh code is of power 2, i.e. N=2n (n is any positive integer). The matrix contain one row of all zeros and the other rows each have equal number of ones and zeros. Walsh codes can be generated by following recursive procedure:

$$W^1 = [0]$$

$$W^2 = \begin{bmatrix} 0 & 0 \\ 0 & 1 \end{bmatrix}$$

$$W^4 = \begin{bmatrix} 0 & 0 & 0 & 0 \\ 0 & 1 & 0 & 1 \\ 0 & 0 & 1 & 1 \\ 0 & 1 & 1 & 0 \end{bmatrix}$$

i.e.

$$W^{2n} = \begin{bmatrix} W^n & W^n \\ W^n & \overline{W^n} \end{bmatrix}$$

Where, N is a power of 2 and over-score implies the binary complement of corresponding bits in the matrix. So, N length Walsh code can provide N number of codes which can serve maximum N number of users. These codes are orthogonal to each other and thus have zero cross-correlation between any pair.

VII. Implementation

The models for the each Direct Sequence and Frequency Hop Spread Spectrum are designed and simulated. MATLAB is the tool in which all the simulation is done and the graphs are plotted. The DSSS system is implemented using BPSK modulator. The m-sequences, Gold Sequence and the Walsh code are generated and used as the PN sequences at the input of the DSSS system. The signal is transmitted through AWGN channel. Finally the BER for each case is calculated. The FHSS is also implemented using BPSK modulator. The modulated signal is then derived by each the PN sequence, Gold Sequence and the Walsh Code. The signal is then transmitted along a channel with AWGN. Also the BER of the FHSS for each case is plotted.

VIII. Simulation Results

The DSSS and FHSS transmitter and receiver is simulated for each m-sequence, GOLD, and Walsh Sequence using the MATLAB in the presence of the noise. The various waveforms are obtained during the different stages. Finally the BER(bit error rate) is plotted for each case.

Fig. 1, Fig.2, Fig.3 shows the PN Sequence, Gold Sequence and Walsh Codes respectively. These are the three types of codes that are used for the simulation and are generated using blocksets.

Fig. 4 and Fig. 5 shows the transmitted and received signal during the DSSS system using PN sequences. The signals indicates itself that it appears as noise during the transmission.

Similarly, Fig. 6, Fig.7 and Fig. 8, Fig. 9 shows the transmitted and received signal during the DSSS system using Gold sequences and Walsh Codes respectively.

Fig. 10 shows the BER (bit error rate) of DSSS using all the codes. The bit error rate remains same for all the used codes in DSSS.

Fig. 11,12, Fig. 13,14 and Fig. 15,16 shows the transmitted and received signals of FHSS using PN sequence, Gold Sequences, and Walsh Codes respectively.

Fig. 17 is the BER of FHSS of PN sequence, Gold Sequence, and Walsh codes.

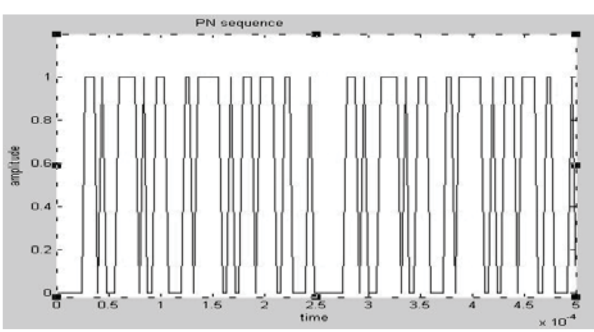

Fig.1 : PN sequence

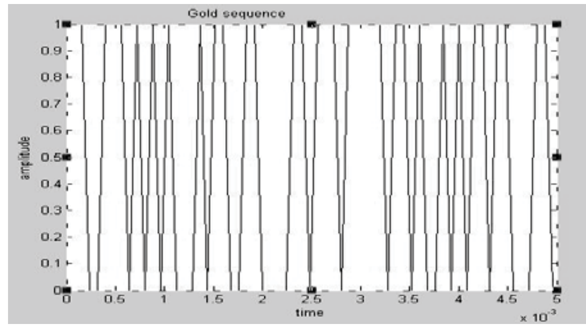

Fig. 2 : Gold Sequence

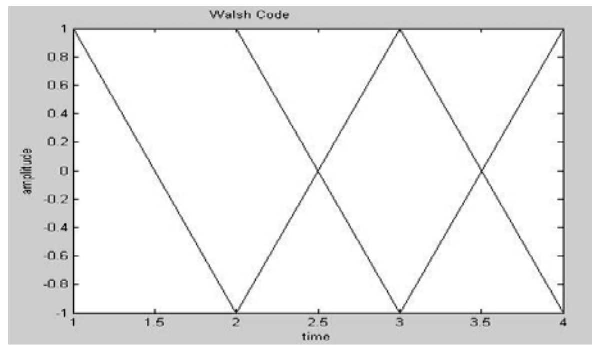

Fig. 3 : Walsh Code

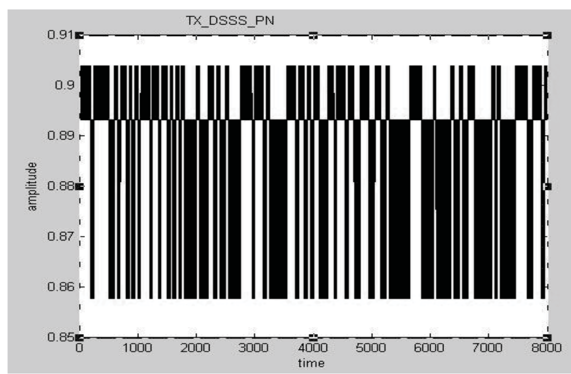

Fig. 4 : DSSS transmitter signal for PN sequence

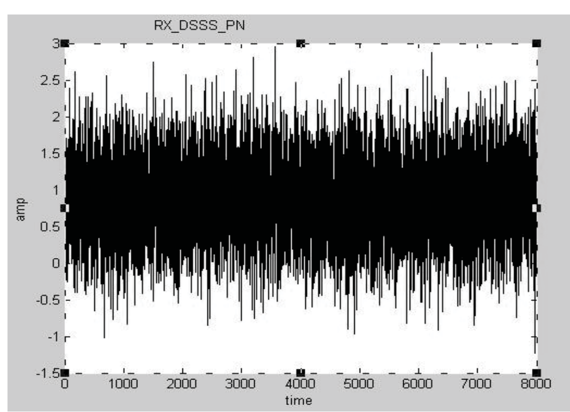

Fig. 5 : DSSS Receivers signal for PN sequence

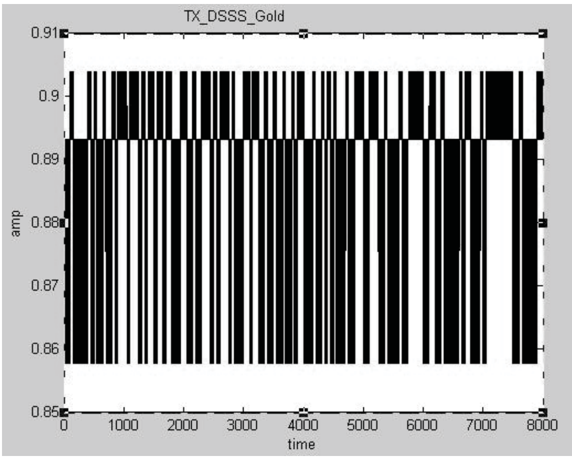

Fig. 6 : DSSS transmitter signal for Gold sequence

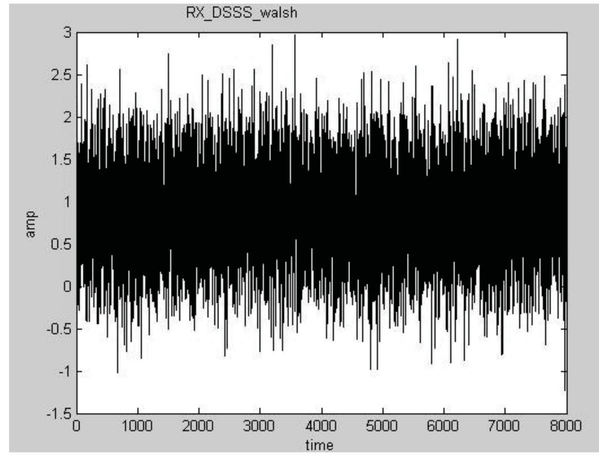

Fig. 9 : DSSS Receivers signal for Walsh sequence

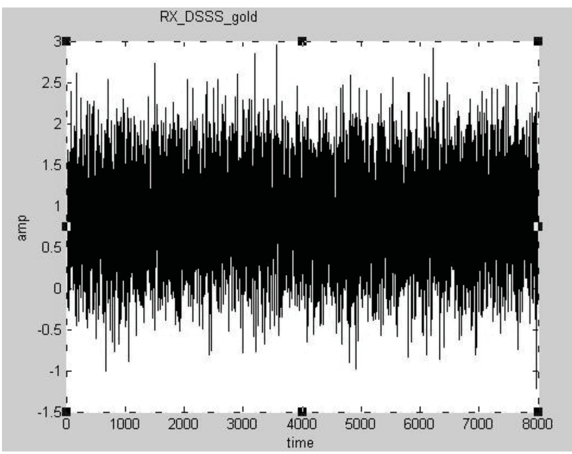

Fig. 7 : DSSS receiver signal for Gold sequence

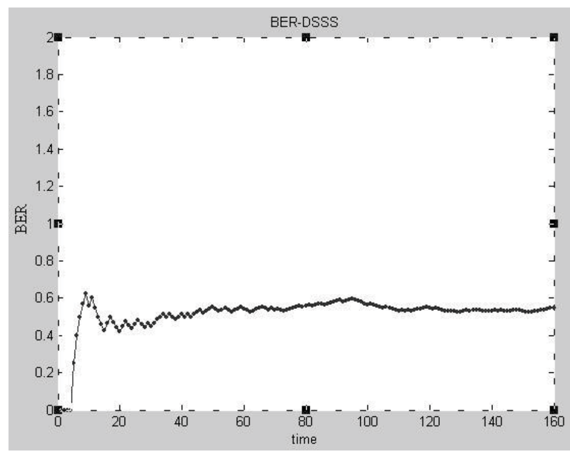

Fig. 10 : BER for DSSS

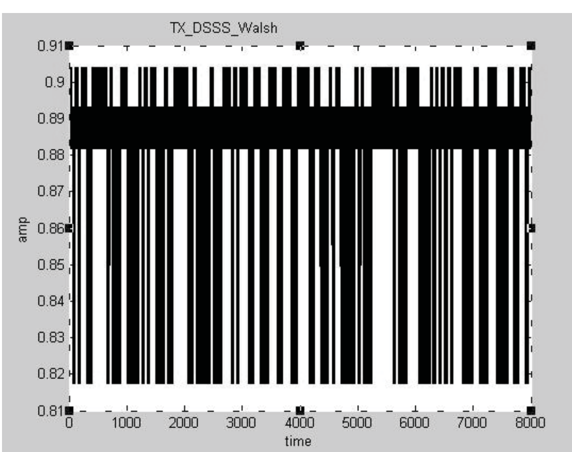

Fig. 8 : DSSS transmitter signal for Walsh

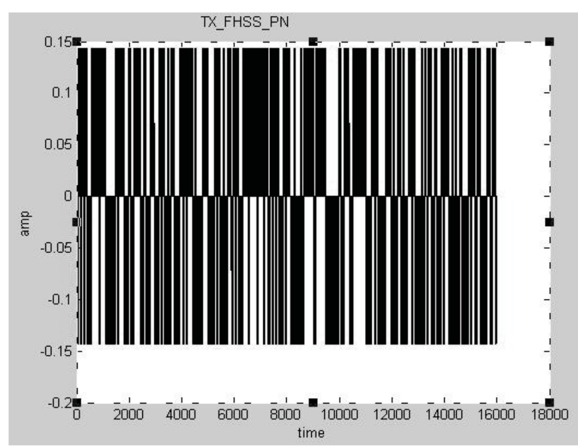

Fig. 11 : FHSS transmitter signal for PN sequence

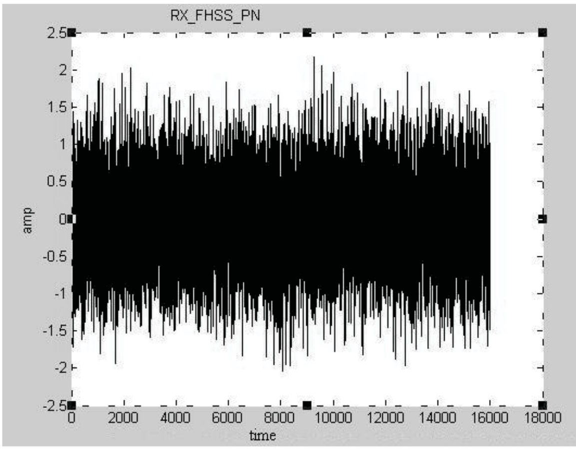

Fig. 12 : FHSS Receiver signal for PN sequence

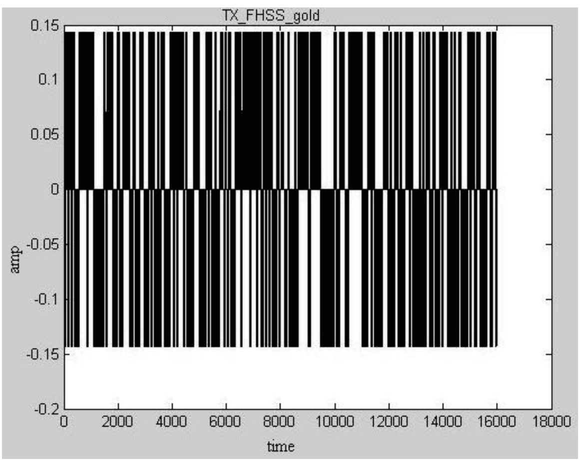

Fig. 13 : FHSS transmitter signal for Gold sequence

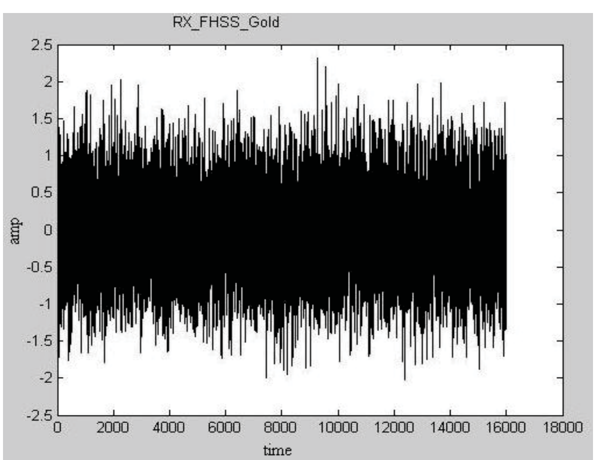

Fig. 14 : FHSS Receiver signal for Gold sequence

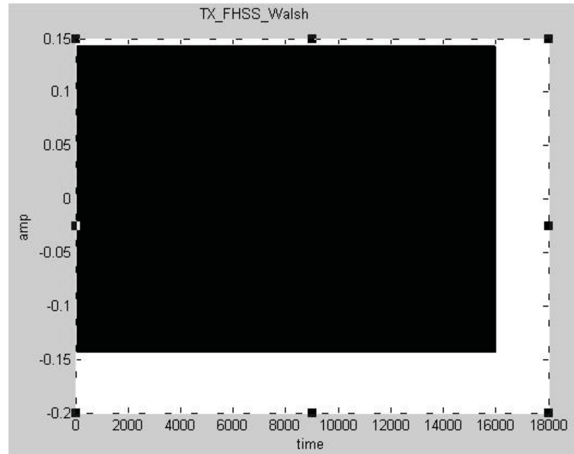

Fig. 15 : FHSS transmitter signal for Walsh

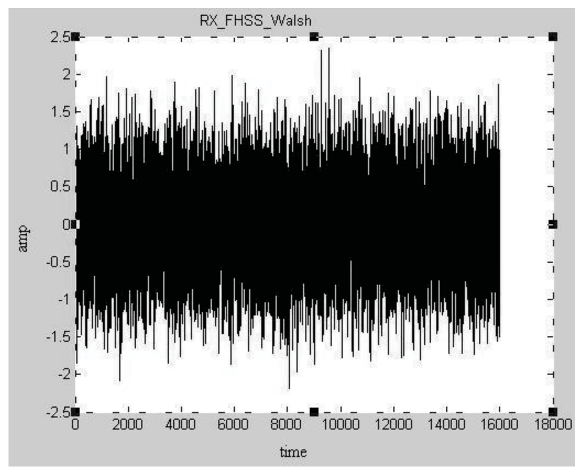

Fig. 16 : FHSS Receiver signal for Walsh

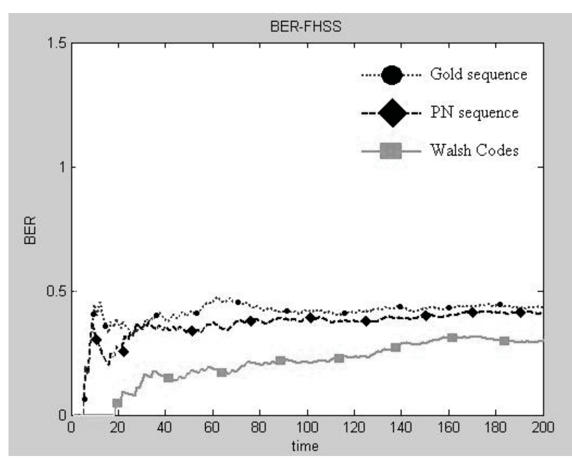

Fig. 17 : BER for FHSS

IX. Conclusion

The result of the MATLAB simulation of the Direct Sequence Spread Spectrum (DSSS) system shows that the if the different codes (i.e. m sequences, Gold Codes, Walsh in this case) are used to implement the DSSS system there is no effect on the BER(bit error rate) of the system. The MATLAB simulation of the Frequency Hop Spread Spectrum (FHSS) system shows that the if the different codes are used to implement the FHSS system the PN sequences shows the large BER. This is due to the non orthogonality of the codes. Whereas the orthogonal code i.e. the Walsh code shows the least BER among all three codes in FHSS system. Walsh code gives the much better performance in FHSS as it shows the less Bit error rate if walsh code is used in place of PN sequences.

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5. Ask your Guides: If you are having any difficulty in your research, then do not hesitate to share your difficulty to your guide (if you have any). They will surely help you out and resolve your doubts. If you can't clarify what exactly you require for your work then ask the supervisor to help you with the alternative. He might also provide you the list of essential readings.

6. Use of computer is recommended: As you are doing research in the field of Computer Science, then this point is quite obvious.

7. Use right software: Always use good quality software packages. If you are not capable to judge good software then you can lose quality of your paper unknowingly. There are various software programs available to help you, which you can get through Internet.

8. Use the Internet for help: An excellent start for your paper can be by using the Google. It is an excellent search engine, where you can have your doubts resolved. You may also read some answers for the frequent question how to write my research paper or find model research paper. From the internet library you can download books. If you have all required books make important reading selecting and analyzing the specified information. Then put together research paper sketch out.

9. Use and get big pictures: Always use encyclopedias, Wikipedia to get pictures so that you can go into the depth.

10. Bookmarks are useful: When you read any book or magazine, you generally use bookmarks, right! It is a good habit, which helps to not to lose your continuity. You should always use bookmarks while searching on Internet also, which will make your search easier.

11. Revise what you wrote: When you write anything, always read it, summarize it and then finalize it.

12. Make all efforts: Make all efforts to mention what you are going to write in your paper. That means always have a good start. Try to mention everything in introduction, that what is the need of a particular research paper. Polish your work by good skill of writing and always give an evaluator, what he wants.

13. Have backups: When you are going to do any important thing like making research paper, you should always have backup copies of it either in your computer or in paper. This will help you to not to lose any of your important.

14. Produce good diagrams of your own: Always try to include good charts or diagrams in your paper to improve quality. Using several and unnecessary diagrams will degrade the quality of your paper by creating "hotchpotch." So always, try to make and include those diagrams, which are made by your own to improve readability and understandability of your paper.

15. Use of direct quotes: When you do research relevant to literature, history or current affairs then use of quotes become essential but if study is relevant to science then use of quotes is not preferable.

16. Use proper verb tense: Use proper verb tenses in your paper. Use past tense, to present those events that happened. Use present tense to indicate events that are going on. Use future tense to indicate future happening events. Use of improper and wrong tenses will confuse the evaluator. Avoid the sentences that are incomplete.

17. Never use online paper: If you are getting any paper on Internet, then never use it as your research paper because it might be possible that evaluator has already seen it or maybe it is outdated version.

18. **Pick a good study spot:** To do your research studies always try to pick a spot, which is quiet. Every spot is not for studies. Spot that suits you choose it and proceed further.

19. Know what you know: Always try to know, what you know by making objectives. Else, you will be confused and cannot achieve your target.

20. Use good quality grammar: Always use a good quality grammar and use words that will throw positive impact on evaluator. Use of good quality grammar does not mean to use tough words, that for each word the evaluator has to go through dictionary. Do not start sentence with a conjunction. Do not fragment sentences. Eliminate one-word sentences. Ignore passive voice. Do not ever use a big word when a diminutive one would suffice. Verbs have to be in agreement with their subjects. Prepositions are not expressions to finish sentences with. It is incorrect to ever divide an infinitive. Avoid clichés like the disease. Also, always shun irritating alliteration. Use language that is simple and straight forward. put together a neat summary.

21. Arrangement of information: Each section of the main body should start with an opening sentence and there should be a changeover at the end of the section. Give only valid and powerful arguments to your topic. You may also maintain your arguments with records.

22. Never start in last minute: Always start at right time and give enough time to research work. Leaving everything to the last minute will degrade your paper and spoil your work.

23. Multitasking in research is not good: Doing several things at the same time proves bad habit in case of research activity. Research is an area, where everything has a particular time slot. Divide your research work in parts and do particular part in particular time slot.

24. Never copy others' work: Never copy others' work and give it your name because if evaluator has seen it anywhere you will be in trouble.

25. Take proper rest and food: No matter how many hours you spend for your research activity, if you are not taking care of your health then all your efforts will be in vain. For a quality research, study is must, and this can be done by taking proper rest and food.

26. Go for seminars: Attend seminars if the topic is relevant to your research area. Utilize all your resources.

27. Refresh your mind after intervals: Try to give rest to your mind by listening to soft music or by sleeping in intervals. This will also improve your memory.

28. Make colleagues: Always try to make colleagues. No matter how sharper or intelligent you are, if you make colleagues you can have several ideas, which will be helpful for your research.

29. **Think technically:** Always think technically. If anything happens, then search its reasons, its benefits, and demerits.

30. Think and then print: When you will go to print your paper, notice that tables are not be split, headings are not detached from their descriptions, and page sequence is maintained.

31. Adding unnecessary information: Do not add unnecessary information, like, I have used MS Excel to draw graph. Do not add irrelevant and inappropriate material. These all will create superfluous. Foreign terminology and phrases are not apropos. One should NEVER take a broad view. Analogy in script is like feathers on a snake. Not at all use a large word when a very small one would be sufficient. Use words properly, regardless of how others use them. Remove quotations. Puns are for kids, not grunt readers. Amplification is a billion times of inferior quality than sarcasm.

32. Never oversimplify everything: To add material in your research paper, never go for oversimplification. This will definitely irritate the evaluator. Be more or less specific. Also too, by no means, ever use rhythmic redundancies. Contractions aren't essential and shouldn't be there used. Comparisons are as terrible as clichés. Give up ampersands and abbreviations, and so on. Remove commas, that are, not necessary. Parenthetical words however should be together with this in commas. Understatement is all the time the complete best way to put onward earth-shaking thoughts. Give a detailed literary review.

33. Report concluded results: Use concluded results. From raw data, filter the results and then conclude your studies based on measurements and observations taken. Significant figures and appropriate number of decimal places should be used. Parenthetical remarks are prohibitive. Proofread carefully at final stage. In the end give outline to your arguments. Spot out perspectives of further study of this subject. Justify your conclusion by at the bottom of them with sufficient justifications and examples.

34. After conclusion: Once you have concluded your research, the next most important step is to present your findings. Presentation is extremely important as it is the definite medium though which your research is going to be in print to the rest of the crowd. Care should be taken to categorize your thoughts well and present them in a logical and neat manner. A good quality research paper format is essential because it serves to highlight your research paper and bring to light all necessary aspects in your research.

Informal Guidelines of Research Paper Writing

Key points to remember:

- Submit all work in its final form.
- Write your paper in the form, which is presented in the guidelines using the template.
- Please note the criterion for grading the final paper by peer-reviewers.

Final Points:

A purpose of organizing a research paper is to let people to interpret your effort selectively. The journal requires the following sections, submitted in the order listed, each section to start on a new page.

The introduction will be compiled from reference matter and will reflect the design processes or outline of basis that direct you to make study. As you will carry out the process of study, the method and process section will be constructed as like that. The result segment will show related statistics in nearly sequential order and will direct the reviewers next to the similar intellectual paths throughout the data that you took to carry out your study. The discussion section will provide understanding of the data and projections as to the implication of the results. The use of good quality references all through the paper will give the effort trustworthiness by representing an alertness of prior workings.

Writing a research paper is not an easy job no matter how trouble-free the actual research or concept. Practice, excellent preparation, and controlled record keeping are the only means to make straightforward the progression.

General style:

Specific editorial column necessities for compliance of a manuscript will always take over from directions in these general guidelines.

To make a paper clear

· Adhere to recommended page limits

Mistakes to evade

- Insertion a title at the foot of a page with the subsequent text on the next page

- Separating a table/chart or figure - impound each figure/table to a single page
- Submitting a manuscript with pages out of sequence

In every sections of your document

· Use standard writing style including articles ("a", "the," etc.)

· Keep on paying attention on the research topic of the paper

· Use paragraphs to split each significant point (excluding for the abstract)

· Align the primary line of each section

· Present your points in sound order

· Use present tense to report well accepted

· Use past tense to describe specific results

· Shun familiar wording, don't address the reviewer directly, and don't use slang, slang language, or superlatives

· Shun use of extra pictures - include only those figures essential to presenting results

Title Page:

Choose a revealing title. It should be short. It should not have non-standard acronyms or abbreviations. It should not exceed two printed lines. It should include the name(s) and address (es) of all authors.

Abstract:

The summary should be two hundred words or less. It should briefly and clearly explain the key findings reported in the manuscript--must have precise statistics. It should not have abnormal acronyms or abbreviations. It should be logical in itself. Shun citing references at this point.

An abstract is a brief distinct paragraph summary of finished work or work in development. In a minute or less a reviewer can be taught the foundation behind the study, common approach to the problem, relevant results, and significant conclusions or new questions.

Write your summary when your paper is completed because how can you write the summary of anything which is not yet written? Wealth of terminology is very essential in abstract. Yet, use comprehensive sentences and do not let go readability for briefness. You can maintain it succinct by phrasing sentences so that they provide more than lone rationale. The author can at this moment go straight to shortening the outcome. Sum up the study, with the subsequent elements in any summary. Try to maintain the initial two items to no more than one ruling each.

- Reason of the study - theory, overall issue, purpose
- Fundamental goal
- To the point depiction of the research
- Consequences, including definite statistics - if the consequences are quantitative in nature, account quantitative data; results of any numerical analysis should be reported
- Significant conclusions or questions that track from the research(es)

Approach:

- Single section, and succinct
- As a outline of job done, it is always written in past tense
- A conceptual should situate on its own, and not submit to any other part of the paper such as a form or table
- Center on shortening results - bound background information to a verdict or two, if completely necessary
- What you account in an conceptual must be regular with what you reported in the manuscript
- Exact spelling, clearness of sentences and phrases, and appropriate reporting of quantities (proper units, important statistics) are just as significant in an abstract as they are anywhere else

Introduction:

The **Introduction** should "introduce" the manuscript. The reviewer should be presented with sufficient background information to be capable to comprehend and calculate the purpose of your study without having to submit to other works. The basis for the study should be offered. Give most important references but shun difficult to make a comprehensive appraisal of the topic. In the introduction, describe the problem visibly. If the problem is not acknowledged in a logical, reasonable way, the reviewer will have no attention in your result. Speak in common terms about techniques used to explain the problem, if needed, but do not present any particulars about the protocols here. Following approach can create a valuable beginning:

- Explain the value (significance) of the study
- Shield the model - why did you employ this particular system or method? What is its compensation? You strength remark on its appropriateness from a abstract point of vision as well as point out sensible reasons for using it.
- Present a justification. Status your particular theory (es) or aim(s), and describe the logic that led you to choose them.
- Very for a short time explain the tentative propose and how it skilled the declared objectives.

Approach:

- Use past tense except for when referring to recognized facts. After all, the manuscript will be submitted after the entire job is done.
- Sort out your thoughts; manufacture one key point with every section. If you make the four points listed above, you will need a least of four paragraphs.
- Present surroundings information only as desirable in order hold up a situation. The reviewer does not desire to read the whole thing you know about a topic.
- Shape the theory/purpose specifically - do not take a broad view.
- As always, give awareness to spelling, simplicity and correctness of sentences and phrases.

Procedures (Methods and Materials):

This part is supposed to be the easiest to carve if you have good skills. A sound written Procedures segment allows a capable scientist to replacement your results. Present precise information about your supplies. The suppliers and clarity of reagents can be helpful bits of information. Present methods in sequential order but linked methodologies can be grouped as a segment. Be concise when relating the protocols. Attempt for the least amount of information that would permit another capable scientist to spare your outcome but be cautious that vital information is integrated. The use of subheadings is suggested and ought to be synchronized with the results section. When a technique is used that has been well described in another object, mention the specific item describing a way but draw the basic principle while stating the situation. The purpose is to text all particular resources and broad procedures, so that another person may use some or all of the methods in one more study or referee the scientific value of your work. It is not to be a step by step report of the whole thing you did, nor is a methods section a set of orders.

Materials:

- Explain materials individually only if the study is so complex that it saves liberty this way.
- Embrace particular materials, and any tools or provisions that are not frequently found in laboratories.
- Do not take in frequently found.
- If use of a definite type of tools.
- Materials may be reported in a part section or else they may be recognized along with your measures.

Methods:

- Report the method (not particulars of each process that engaged the same methodology)
- Describe the method entirely
- To be succinct, present methods under headings dedicated to specific dealings or groups of measures
- Simplify - details how procedures were completed not how they were exclusively performed on a particular day.
- If well known procedures were used, account the procedure by name, possibly with reference, and that's all.

Approach:

- It is embarrassed or not possible to use vigorous voice when documenting methods with no using first person, which would focus the reviewer's interest on the researcher rather than the job. As a result when script up the methods most authors use third person passive voice.
- Use standard style in this and in every other part of the paper - avoid familiar lists, and use full sentences.

What to keep away from

- Resources and methods are not a set of information.
- Skip all descriptive information and surroundings - save it for the argument.
- Leave out information that is immaterial to a third party.

Results:

The principle of a results segment is to present and demonstrate your conclusion. Create this part a entirely objective details of the outcome, and save all understanding for the discussion.

The page length of this segment is set by the sum and types of data to be reported. Carry on to be to the point, by means of statistics and tables, if suitable, to present consequences most efficiently.You must obviously differentiate material that would usually be incorporated in a study editorial from any unprocessed data or additional appendix matter that would not be available. In fact, such matter should not be submitted at all except requested by the instructor.

Content

- Sum up your conclusion in text and demonstrate them, if suitable, with figures and tables.
- In manuscript, explain each of your consequences, point the reader to remarks that are most appropriate.
- Present a background, such as by describing the question that was addressed by creation an exacting study.
- Explain results of control experiments and comprise remarks that are not accessible in a prescribed figure or table, if appropriate.
- Examine your data, then prepare the analyzed (transformed) data in the form of a figure (graph), table, or in manuscript form.

What to stay away from

- Do not discuss or infer your outcome, report surroundings information, or try to explain anything.
- Not at all, take in raw data or intermediate calculations in a research manuscript.

- Do not present the similar data more than once.
- Manuscript should complement any figures or tables, not duplicate the identical information.
- Never confuse figures with tables - there is a difference.

Approach

- As forever, use past tense when you submit to your results, and put the whole thing in a reasonable order.
- Put figures and tables, appropriately numbered, in order at the end of the report
- If you desire, you may place your figures and tables properly within the text of your results part.

Figures and tables

- If you put figures and tables at the end of the details, make certain that they are visibly distinguished from any attach appendix materials, such as raw facts
- Despite of position, each figure must be numbered one after the other and complete with subtitle
- In spite of position, each table must be titled, numbered one after the other and complete with heading
- All figure and table must be adequately complete that it could situate on its own, divide from text

Discussion:

The Discussion is expected the trickiest segment to write and describe. A lot of papers submitted for journal are discarded based on problems with the Discussion. There is no head of state for how long a argument should be. Position your understanding of the outcome visibly to lead the reviewer through your conclusions, and then finish the paper with a summing up of the implication of the study. The purpose here is to offer an understanding of your results and hold up for all of your conclusions, using facts from your research and generally accepted information, if suitable. The implication of result should be visibly described. Infer your data in the conversation in suitable depth. This means that when you clarify an observable fact you must explain mechanisms that may account for the observation. If your results vary from your prospect, make clear why that may have happened. If your results agree, then explain the theory that the proof supported. It is never suitable to just state that the data approved with prospect, and let it drop at that.

- Make a decision if each premise is supported, discarded, or if you cannot make a conclusion with assurance. Do not just dismiss a study or part of a study as "uncertain."
- Research papers are not acknowledged if the work is imperfect. Draw what conclusions you can based upon the results that you have, and take care of the study as a finished work
- You may propose future guidelines, such as how the experiment might be personalized to accomplish a new idea.
- Give details all of your remarks as much as possible, focus on mechanisms.
- Make a decision if the tentative design sufficiently addressed the theory, and whether or not it was correctly restricted.
- Try to present substitute explanations if sensible alternatives be present.
- One research will not counter an overall question, so maintain the large picture in mind, where do you go next? The best studies unlock new avenues of study. What questions remain?
- Recommendations for detailed papers will offer supplementary suggestions.

Approach:

- When you refer to information, differentiate data generated by your own studies from available information
- Submit to work done by specific persons (including you) in past tense.
- Submit to generally acknowledged facts and main beliefs in present tense.

Administration Rules Listed Before
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Please carefully note down following rules and regulation before submitting your Research Paper to Global Journals Inc. (US):

Segment Draft and Final Research Paper: You have to strictly follow the template of research paper. If it is not done your paper may get rejected.

- The **major constraint** is that you must independently make all content, tables, graphs, and facts that are offered in the paper. You must write each part of the paper wholly on your own. The Peer-reviewers need to identify your own perceptive of the concepts in your own terms. NEVER extract straight from any foundation, and never rephrase someone else's analysis.

- Do not give permission to anyone else to "PROOFREAD" your manuscript.

- Methods to avoid Plagiarism is applied by us on every paper, if found guilty, you will be blacklisted by all of our collaborated research groups, your institution will be informed for this and strict legal actions will be taken immediately.)
- To guard yourself and others from possible illegal use please do not permit anyone right to use to your paper and files.

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Topics	Grades		
	A-B	**C-D**	**E-F**
Abstract	Clear and concise with appropriate content, Correct format. 200 words or below	Unclear summary and no specific data, Incorrect form Above 200 words	No specific data with ambiguous information Above 250 words
Introduction	Containing all background details with clear goal and appropriate details, flow specification, no grammar and spelling mistake, well organized sentence and paragraph, reference cited	Unclear and confusing data, appropriate format, grammar and spelling errors with unorganized matter	Out of place depth and content, hazy format
Methods and Procedures	Clear and to the point with well arranged paragraph, precision and accuracy of facts and figures, well organized subheads	Difficult to comprehend with embarrassed text, too much explanation but completed	Incorrect and unorganized structure with hazy meaning
Result	Well organized, Clear and specific, Correct units with precision, correct data, well structuring of paragraph, no grammar and spelling mistake	Complete and embarrassed text, difficult to comprehend	Irregular format with wrong facts and figures
Discussion	Well organized, meaningful specification, sound conclusion, logical and concise explanation, highly structured paragraph reference cited	Wordy, unclear conclusion, spurious	Conclusion is not cited, unorganized, difficult to comprehend
References	Complete and correct format, well organized	Beside the point, Incomplete	Wrong format and structuring

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