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# Editorial Preface

It is a great privilege for me as Editor in Chief of International Journal of Logic and Computation (IJLP) to present our readers the current issue of Journal which wraps up its first year and first issue of successful publication. This journal has focused on publishing research that provides information for practitioners, researchers and academicians with a teaching or research interest in engineering and science discipline. The first issue of IJLP is organized to presents articles in a particular area of computer logic and computation to attract readers who are interested in reading papers related to that special field. The first issue of IJLP provides a better chance to fulfill the anticipation of a broader community of our audiences.

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Professor. Santosh Kumar Nanda
**Editor-In-Chief**
International Journal of Logic and Computation (IJLP)

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# Design of an Adaptive Hearing Aid Algorithm using Booth-Wallace Tree Multiplier

**Jitendra Kumar Das** jkdas12@gmail.com

*Asst. Professor, Dept. of Electronics and Telecommunication Engineering, Synergy Institute of Engineering & Technology, Dhenkanal, 759001, Orissa, India*

**Dr. K. K. Mahapatra** kmaha2@rediffmail.com

*Professor, Dept. of Electronics and Communication, NIT Rourkela, 769008, Orissa, India*

**Abstract**

The paper presents FPGA implementation of a spectral sharpening process suitable for speech enhancement and noise reduction algorithms for digital hearing aids. Booth and Booth Wallace multiplier is used for implementing digital signal processing algorithms in hearing aids. VHDL simulation results confirm that Booth Wallace multiplier is hardware efficient and performs faster than Booth's multiplier. Booth Wallace multiplier consumes 40% less power compared to Booth multiplier. A novel digital hearing aid using spectral sharpening filter employing booth Wallace multiplier is proposed. The results reveal that the hardware requirement for implementing hearing aid using Booth Wallace multiplier is less when compared with that of a booth multiplier. Furthermore it is also demonstrated that digital hearing aid using Booth Wallace multiplier consumes less power and performs better in terms of speed.

**Keywords:** Booth Multiplier, Booth Wallace Multiplier, Adaptive Lattice Filte

## 1. INTRODUCTION

The decimation filter used in designing of a hearing aid has two major disadvantages which are

a. It requires more area for designing in FPGA.
b. As it is highly serial in nature, it requires more output latency.

Due to above reasons it consumes more power and we are specifically interested on lowering the power consumption of digital hearing aids. In this context we have used our own customized multiplier while maintaining the overall signal quality [6-7] to lower the power consumption.

Hearing impairment is often accompanied with reduced frequency selectivity which leads to a decreased speech intelligibility in noisy environments [2-5]. One possibility to alleviate this deficiency is the spectral sharpening for speech enhancement based on adaptive filtering [8-9] by which the intelligibility of the speech signal is maintained. Due to area constraints, such algorithms are usually implemented in totally time-multiplexed architectures, in which multiple operations are scheduled to run on a few processing units. This work discusses the power

consumption in an FPGA implementation of the speech enhancement algorithm. It points out that power consumption can be reduced using Booth Wallace multiplier [11]. Several implementations of the algorithm, differing only in the degree of resource sharing are investigated aiming at power-efficiency maximization. At first an overview of the algorithm is given. Next the realized architectures using booth-Wallace tree multiplier are presented.

## 2. WALLACE TREE MULTIPLIER

Wallace trees are irregular in the sense that the informal description does not specify a systematic method for the compressor interconnections [10]. However, it is an efficient implementation of adding partial products in parallel [12]. The Wallace tree operates in three steps:

- Multiply: The multiplication is carried out using Booth Algorithm [11-14] which will generate n/2 partial product where n is number of bits of the multiplicand. The partial products are generated using the Booth recoding table given in table 1.

| $Mr_{i+1}$ | Mr | $Mr_{i-1}$ | Recoded output |
|---|---|---|---|
| 0 | 0 | 0 | 0 |
| 0 | 0 | 1 | Y |
| 0 | 1 | 0 | Y |
| 0 | 1 | 1 | +2Y |
| 1 | 0 | 0 | -2Y |
| 1 | 0 | 1 | -Y |
| 1 | 1 | 0 | -Y |
| 1 | 1 | 1 | 0 |

**TABLE 1:** Booth recoding table

- Addition: As long as there are more than 3 wires with the same weights add a following layer. Take 3 wires of same weight and input them into a full adder. The result will be an output wire of same weight. If there are two wires of same weight, add them using half-adder and if only one is left, connect it to the next layer.
- Group the wires in two numbers and add in a conventional adder. A typical Wallace tree architecture is shown in figure1 below. In the diagram AB0-AB7 represents the partial products.

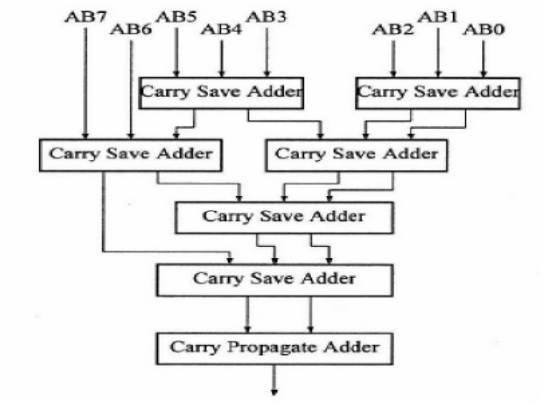


**FIGURE 1:** Wallace tree multiplier

Wallace multipliers consist of AND-gates, Carry Save Adders and a Carry Propagate Adder or Carry Look-ahead Adder.
The n-bit CSA consists of disjoint full adders (FA's). It consumes three-bit input vectors and produces two outputs, i.e., n-bit sum vector S and n-bit carry vector *C*. Unlike the normal adders [e.g. ripple-carry adder (RCA) and carry-look ahead adder (CLA)], a CSA contains no

carry propagation. Consequently, the CSA has the same propagation delay as only one FA delay and the delay is constant for any value of n. For sufficiently large n, the CSA implementation becomes much faster and also relatively smaller in size than the implementation of normal adders. In Wallace multiplier carry save adders are used, and one carry propagate adder is used as shown in the figure 2. The basic idea in Wallace multiplier is that all the partial products are added at the same time instead of adding one at a time. This speeds up the multiplication process.

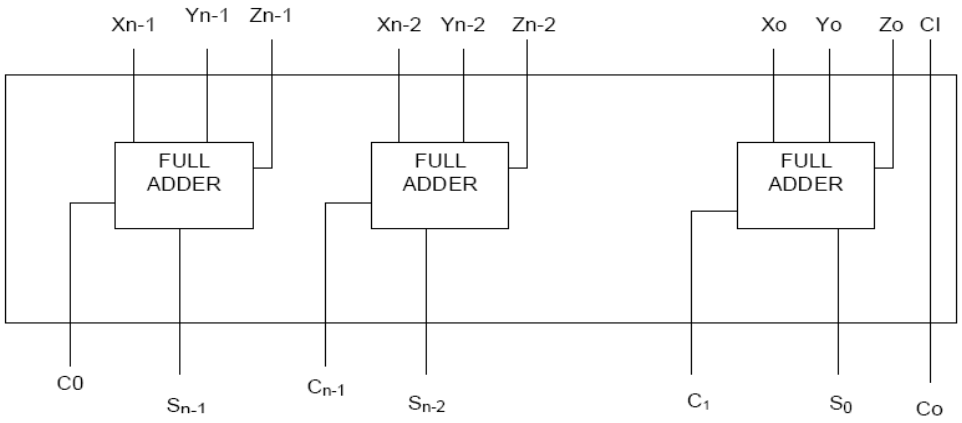


**FIGURE 2:** Implementation of n bit CSA operation

## 3. ADAPTIVE LATTICE FILTER

The adaptive lattice filter [6-7, 18-20] consists of three parts and these are

a. Adaptive decorrelator
b. Analysis filter (lattice filter)
c. Synthesis filter(lattice structure)

**The Adaptive Decorrelator**

An adaptive filter is a filter that adjusts its transfer function according to an optimizing algorithm. Because of the complexity of the optimizing algorithms, most adaptive filters are digital filters that perform digital signal processing and adapt their performance based on the input signal used.

The adaptive process involves the use of a cost function, which is a criterion for optimum performance so that the filter coefficients adjusted to minimize the cost on the next iteration [1]. The block diagram for such filter is presented in figure 3 that serves as a foundation for particular adaptive filter realizations, such as Least Mean Squares (LMS), Recursive Least Squares (RLS) or steepest descent algorithm etc. The idea behind the block diagram is that a variable filter extracts an estimate of the desired signal.

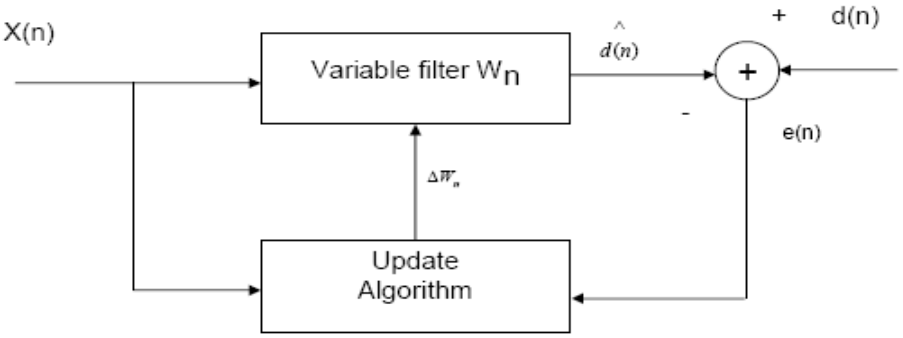


**FIGURE 3:** Block diagram of an Adaptive filter

The structure presented in figure 3 is described now. We take the following assumptions:

1. The input signal is the sum of a desired signal d(n) and interfering noise v(n)

$$x(n) = d(n) + v(n) \quad (1)$$

2. The variable filter has a Finite Impulse Response (FIR) structure. For such structures the impulse response is equal to the filter coefficients. The coefficients for a filter of order p are defined as

$$W_n(0)=[w_n(0),w_n(1),\ldots w_n(p)]^T \quad (2)$$

3. The error signal or cost function is the difference between the desired and the estimated signal.

$$e(n) = d(n) - \hat{d}(n) \quad (4)$$

The variable filter estimates the desired signal by convolving the input signal with the impulse response. In vector notation this is expressed as

$$d(n) = W^T(n)X(n) \quad (5)$$

where

$$x(n)=[x(n), x(n-1),\ldots\ldots\ldots,x(n-p)]^T \quad (6)$$

is an input signal vector. Moreover, the variable filter updates the filter coefficients at every time instant

$$W_{n+1}=W_n+ \Delta W_n \quad (7)$$

where $\Delta w_n$ is a correction factor for the filter coefficients. The adaptive algorithm generates this correction factor based on the input and error signals. LMS and RLS define two different coefficient update algorithms. The speech signal to be transmitted is spectrally masked by noise. By using an adaptive filter, we can attempt to minimize the error by finding the correlation between the noise at the signal microphone and the (correlated) noise at the reference microphone. In this particular case the error does not tend to zero as we note the signal d(k) = x(k) + n(k) whereas the input signal to the filter is x(k) and n(k) does not contain any speech [2]. Therefore it is not possible to "subtract" any speech when forming e(k)=d(k)-d(n) . Hence in minimising the power of the error signal e(k) we note that only the noise is removed and e(k) =~ x(k).

Figure 4 depicts the structure of the adaptive gradient lattice decorrelator. The illustration shows three stages only with indices 1, i and m. good results typically require a filter order m where m can vary from 8 to 10 for speech sampled at 8 kHz. The output signal with vanishing autocorrelation is computed on the upper signal path by subtracting from the input sample suitable fractions of the signal values on the lower path. The multipliers $K_1,K_2\ldots\ldots\ldots K_m$ are iteratively computed as in equation 8.

$$K_i[n]:=K_i[n-1]+\Delta K_i[n] \quad (8)$$

At every sampling interval n. the details of this process are illustrated in figure 4 for the i-th stage. Input and output values on upper and lower signal path to and from the $i^{th}$ stage contribute to the computation of the update value $\Delta K_i$.

Both output values are multiplied with the input values on the opposite path and then summed up to form the numerator in the subsequent computation of the update value. The denominator $\sigma^2$ is iteratively computed.

$$\sigma^2 := e.\ \sigma^2[n-1] + \Delta\sigma^2[n] \qquad (9)$$

The incremental value equals the sum of the two squared input values. The iterative computation of the denominator defines an exponentially decaying window which progressively decreases the influence of past contributions.

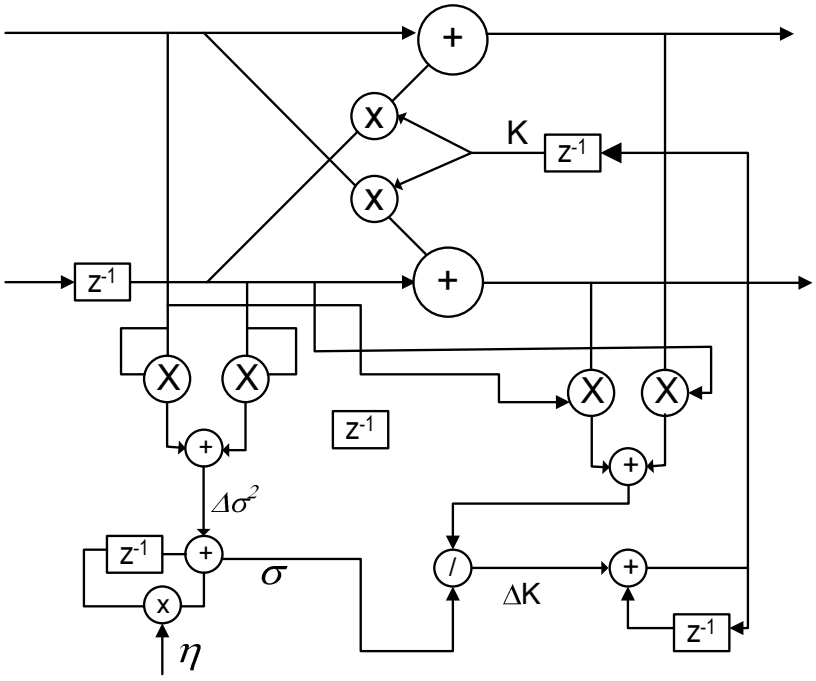


**FIGURE 4:** Adaptive gradient lattice filter

The computationally expensive division yields fast converging filter coefficients $k_i$ independent of the varying input signal power level. This remarkable property is indispensable for good enhancement results. It is also clear contrast to simpler algorithms replacing the division by a multiplication with a small convergence constant $0<\mu<<1$. The longest delay through a string of lattice filters extends from the output of the storage element in the first lattice filter, through a multiplication with the first reflection coefficient, and then through an addition for each stage of the lattice filter until the output is produced in the final stage. For a large number of lattice filter stages, this longest delay can be reduced by a lattice filter optimization for speed which defers the final carry propagating addition until after the final lattice filter stage. This requires the transmission of an additional value between lattice filter stages. The multiplication process is speeded up using booth multiplier and the accumulation process is done faster using the Wallace multiplier.

**The Analysis Filter**

The analysis filter $H(z)=[1-A(z/\beta)]$ is illustrated in figure 5. Its structure [1, 8-9] is similar to that of the adaptive decorrelator shown in figure 4. The only difference is the multiplication with the filter parameter $\beta$ following every shift element $z^{-1}$ on the lower signal path. Furthermore, the analysis filter does not need a separate circuitry for coefficient update. It instead requires and

therefore copies the filter coefficients $K_1$, $K_2$…..$K_m$ computed by the unmodified ($\beta$ =1) filter structure, i.e., the adaptive decorrelator.

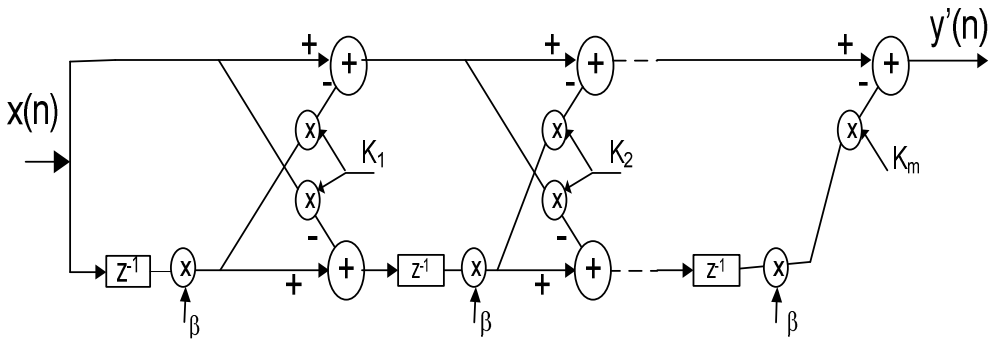


**FIGURE 5:** Analysis filter (1-A(z/□), x(n) is input and y'(n) is output

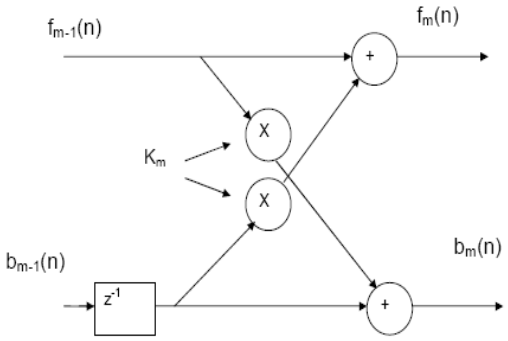


**FIGURE 6:** Single stage of analysis filter

The two mathematical equations for the single stage analysis filter as shown in figure 6 are

$$f_m(n) = f_{m-1}(n) - k_m b_{m-1}(n-1) \quad (10)$$

$$b_m(n) = b_{m-1}(n-1) - k_m f_{m-1}(n) \quad (11)$$

Some characteristics of the Lattice predictor are

- It is the most efficient structure for generating simultaneously the forward and backward prediction errors.
- The lattice structure is modular: increasing the order of the filter requires adding only one extra module, leaving all other modules the same.
- The various stages of a lattice are decoupled from each other in the following sense: The memory of the lattice (storing $b0(n ¡ 1); : : : ; bm¡1(n ¡ 1)$) contains orthogonal variables, thus the information contained in $x(n)$ is splitted in m pieces, which reduces gradually the redundancy of the signal.
- The similar structure of the lattice filter stages makes the filter suitable for VLSI implementation.

Lattice filters typically find use in such applications as predictive filtering, adaptive filtering, and speech processing. One desirable feature of lattice filters are their use of reflection coefficients as the filter parameter. Algorithms exist to compute reflection coefficients to obtain the optimal linear filter for a given filter order. Reflection coefficients have the additional property that for some applications, the optimal reflection coefficients remain unchanged when going from a lower order filter to a higher order filter. Thus, when adding additional filter stages, only the reflection coefficients for the added stages need to be computed. The

modification with filter parameter  causes the analysis filter to produce an output signal with reduced formants instead of a signal with completely flat spectral envelope as produced by the adaptive decorrelator.

**The Synthesis Filter**

When considering IIR filters, the direct form filter is the common structure of choice. This is true, in general, because when designing an algorithm which adapts the parameters $a_k$ and $b_k$, the coefficients of the difference equation, described below, are manipulated directly.

$$Y_k+a_1Y_{k-1}+- - - - - - - - - - - - +a_mY_{k-m}=b_0U_k+b_1U_{k-1}+- - - - - +b_mU_{n-m} \quad (12)$$

Some problems exist in using the direct form filter for adaptive applications. First of all, ensuring stability of a time-varying direct form filter can be a major difficulty. It is often computationally a burden because the polynomial, A(z), made up of the $a_k$ parameters, must be checked to see if it is minimum phase at each iteration. Even if the stability was assured during adaptation, round off error causing limit cycles can plague the filter. Parallel and cascade forms are often used as alternatives for direct form filters. These consist of an interconnection of first and second order filter sections, whose sensitivity to round off errors tends to be less drastic than for the direct form filter. Since the filter is broken down into a factored form, the round off error associated with each factorization only affects that term. In the direct form filter, the factors are lumped together so that round off error in each term affects all of the factors in turn.

A larger problem exists for both parallel and cascade forms: the mapping from transfer function space to parameter space is not unique. Whenever the mapping from the transfer function space to the parameter space is not unique, additional saddle points in the error surface appear that would not be present if the mapping had been unique. The addition of these saddle points can slow down the convergence speed if the parameter trajectories wander close to these saddle points. For this reason, these filter forms are considered unsuitable for adaptive filtering.

A tapped-state lattice form has many of the desirable properties associated with common digital filters and avoids the problems discussed above. Due to the computational structure, the round off error in this filter is inherently low.

Direct implementation of the IIR filter can lead to instabilities if it is quantized. The filter is stable using the following structure [1, 7-9, 14-16]. The structure of the synthesis filter $H(z)=[1-A(z/\gamma)]^{-1}$ is shown in figure 7. The synthesis filter also requires and copies the filter coefficients $K_1$, $K_2$,……$K_m$ from the adaptive decorrelator at every sampling interval. The structure in figure 4.7 also shows a synthesis filter modified by the multiplication with the filter parameter $\gamma$ succeeding every shift element $z^{-1}$ on the lower signal path. The unmodified synthesis filter ($\gamma$=1) restores the original formants in the output when a signal with flat spectral envelope is fed to its input. The modification with a parameter value less than unity causes the synthesis filter to produce an output signal with partially restored formants only. The spectral sharpening effect results from a suitable choice of both filter parameters $0<\beta<\gamma<1$. Experiments with one adaptive filter only failed in producing satisfactory speech enhancement results.

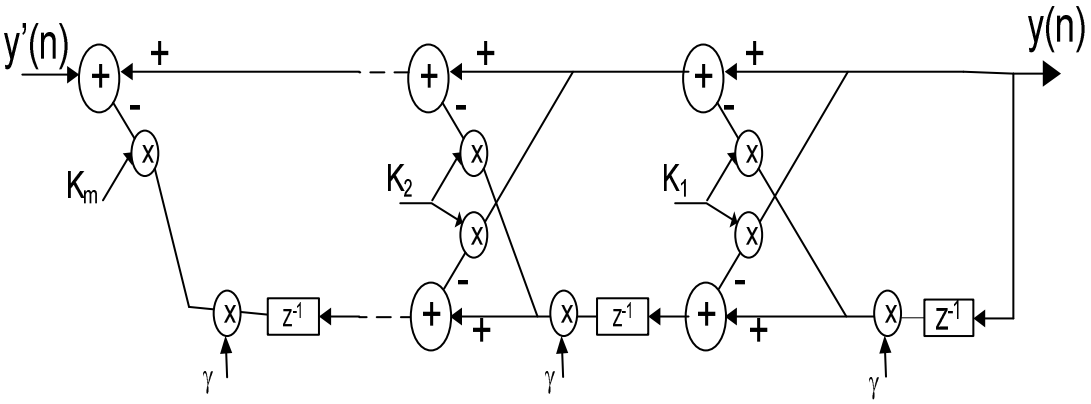


**FIGURE 7:** Synthesis filter, y'(n) input and y(n) is output

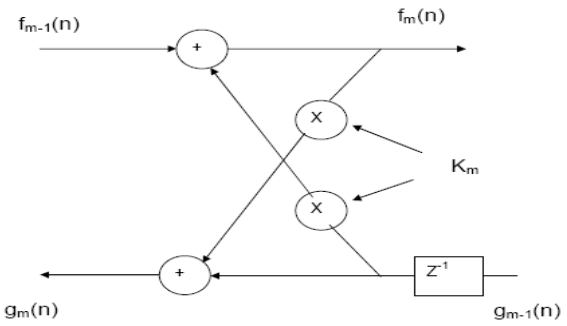


**FIGURE 8:** Single stage of sysnthesi filter.

The two mathematical equations for the single stage synthesis filter are shown below.

$$f_m(n) = f_{m-1}(n)-k_m b_{m-1}(n-1) \quad (13)$$

$$g_m(n) = g_{m-1}(n-1)-k_m f_{m-1}(n) \quad (14)$$

The computational complexity of a digital filter structure is given by the total number of multipliers and the total number of two input adders required for its implementation which roughly provides an indication of its cost of implementation. The synthesis filter is stable if the magnitudes of all multiplier coefficients in the realization are less than unity i.e. - $1<K_m<1$ for m=M, M-1, M-2, …1, 0.

## 4. HEARING AID DESIGN

### 4.1 Spectral Sharpening For Speech Enhancement

Speech enhancement usually results from adaptively filtering the noise reference signals and subsequently subtracting them from the primary input. However, a procedure for speech enhancement based on a single audio path is presented here. It is therefore applicable for real world situations. An example of such a situation is using hearing aid equipment. The hearing impaired person could place additional microphones close to noise sources only rarely. Current hearing aid equipment are used for filtering and amplifying the speech signal, this suggests that hearing impairment is just a more or less reduced sensitivity to sound pressure in various frequency intervals. This view however neglects the loss of frequency discrimination which can be efficiently compensated by the spectral sharpening technique presented. The idea of spectral sharpening originates from the adaptive post filtering method in modern speech coding schemes at bit rates around 8 kb/s and lower [1]. With these algorithms speech is encoded segment by segment. The linear prediction filter is any way computed in every speech segment for the encoding process as

$$A(z)=a_1 z^{-1} + a_2 z^{-2}+.....+ a_m z^{-m} \quad (15)$$

and post- filtering with the transfer function

$$H(z) = \frac{1 - A(z/\beta)}{1 - A(z/\gamma)} \tag{16}$$

and constant filter parameters $0<\beta<\gamma<1$ is subsequently performed with a moderate computational increase.

Figure 9 shows the block diagram of spectral sharpening of speech sharpening [8-9] for speech enhancement. The speech signal x[n] from the microphone splits into three distinct paths. The signal on the lowest path passes through the analysis filter [1-A(z/β)] and subsequently through the synthesis filter $[1-A(z/\gamma)]^{-1}$. Both filters are implemented as lattice filters with the analysis and synthesis structures respectively. They both require the identical set of reflection coefficients $K_1$, $K_2$,.....$K_m$. where m represents the number of stages which is updated in every sampling interval by the adaptive decorrelator shown on the middle path of figure 4. The filter parameters β and γ do not vary with time.

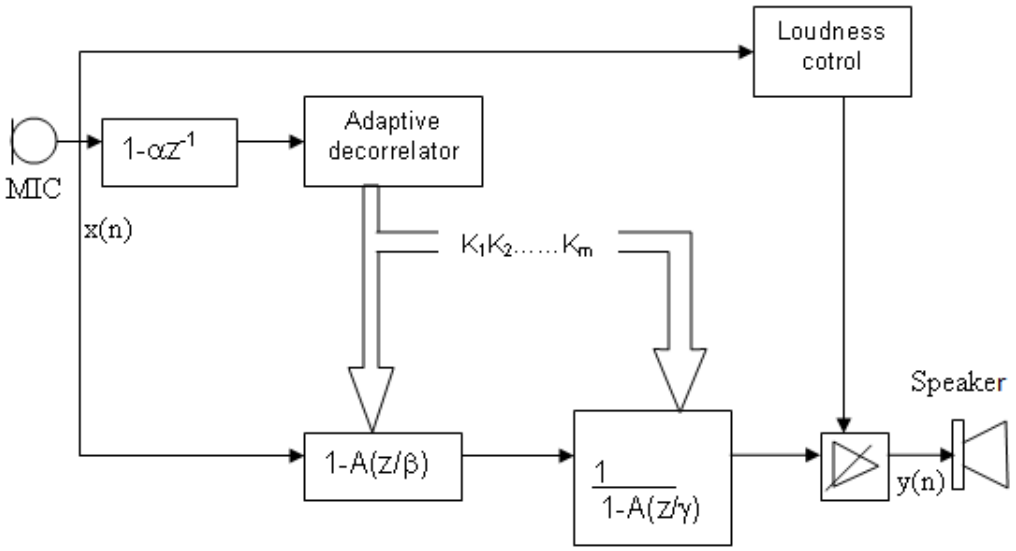


**FIGURE 9:** Block diagram of Spectral Sharpening for Speech Enhancement.

A high pass filter $1-\alpha z^{-1}$ is shown in front of the adaptive decorrelator, where x=1 may be chosen for simplicity. The high pass filter is used in order to compensate the spectral tilt of natural speech: the average power of the speech signal decreases above 1 KHz at a rate of ~ 10 db per octave. The adaptive transfer function in equation (16) enhances this spectral tilt even more when the filter coefficients $K_1$,$K_2$,……..$K_m$ are computed from the speech signal x[n] directly. Efficient speech enhancement requires however that the various formants are more or less uniformly emphasized, regardless of their relative power level. This is possible with the use of the high pass filter. It compensates at least partially the original spectral tilt.

The decorrelator on the middle signal path of the figure is an adaptive gradient lattice filter. It produces an output signal with vanishing autocorrelation by updating its filter coefficients in every sampling interval to the continuously changing input signal characteristics. The output signal is not required in this application, however. The updated filter coefficients $K_1$, $K_2$……$K_m$ are of interest only for the use in the analysis and synthesis filter.

### 4.2 Spectral Sharpening For Noise Reduction

The block diagram of the spectral sharpening process for noise reduction is illustrated in figure 10. The arrangement of adaptive decorrelator, analysis and synthesis filters agrees with the previous block diagram in figure 9, however there various differences like

1. no loudness control,.
2. the input signal x[n] goes directly to the adaptive decorrelator, and

3. a high pass filter precedes the analysis and synthesis filters.

$$H_{hp}(z) = \frac{b(1-z^{-1})}{1-az^{-1}} \tag{17}$$

The reasons for these differences are as follows.
As mentioned in the previous section the spectral sharpening process

$$H(z) = \frac{1-a(z/\beta)}{1-a(z/\gamma)} \tag{18}$$

introduces a signal dependent amplification, signal segments with strong formant structure are amplified more than segments with a rather flat spectral envelop. In the sequel it is assumed that back ground noise is the major source for signal degradation and that its spectrum reveals relatively flat resonances only. Speech segments with strong resonances clearly profit in this situation. They experience a remarkable amplification compared to noisy segments. The loudness compensation of the previous block diagram is consequently omitted in order to preserve this effect.

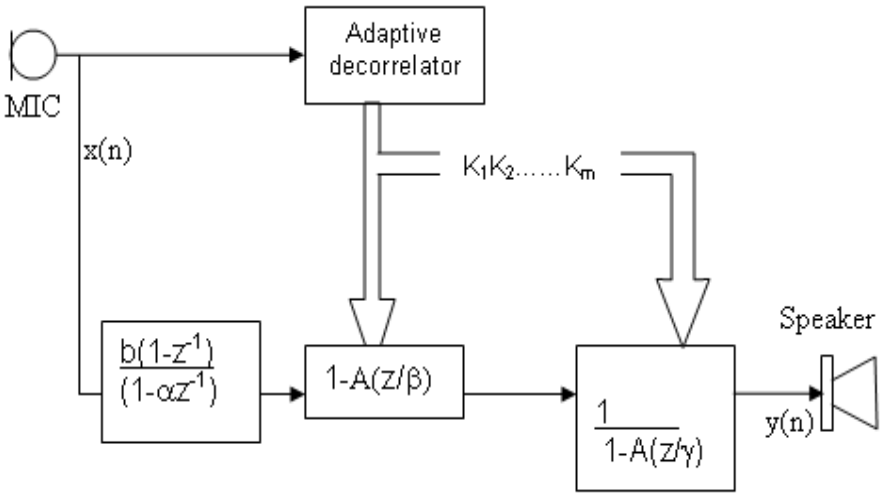


**FIGURE 10:** Block diagram of Spectral Sharpening by Noise Reduction

Best results require that the input signal is directly fed to the adaptive decorrelator. Only negligible amplification is then applied to noisy signal segments as a consequence of their assumed approximately flat spectrum. The spectral sharpening process further enhances the spectral tilt of speech when the filter parameters are estimated from the speech signal without prior compensation.
The high pass filter which preceded the adaptive decorrelator in the figure 9 has been shifted to the bottom signal path in figure 10 in order to avoid the scheme from producing a dull sound.

### 4.3 High Pass Filter

In signal processing, there are many instances in which an input signal to a system contains extra unnecessary content or additional noise which can degrade the quality of the desired signal. In such cases we may remove or filter out the useless samples. For example, in the case of the telephone system, there is no reason to transmit very high frequencies since most speech falls within the band of 700 to 3,400 Hz. Therefore, in this case, all frequencies above and below that band are filtered out. The frequency band between 700 and 3,400 Hz, which isn't filtered out, is known as the pass band, and the frequency band that is blocked out is known as the stop band. FIR, Finite Impulse Response, filters are one of the primary types of filters used in Digital Signal Processing [1, 14-16]. FIR filters are said to be finite because they do not have any feedback. Therefore, if an impulse is sent through the system (a single spike) then the output would invariably become zero as soon as the impulse runs through the filter.

There are a few terms that are used to describe the behaviour and performance of FIR filter. These are

- Filter Coefficients - The set of constants, also called tap weights, used to multiply against delayed sample values. For an FIR filter, the filter coefficients are, by definition, the impulse response of the filter.

- Impulse Response – A filter's time domain output sequence when the input is an impulse. An impulse is a single unity-valued sample followed and preceded by zero valued samples. For an FIR filter the impulse response of a FIR filter is the set of filter coefficients.

- Tap – The number of FIR taps, typically N, tells us a couple things about the filter. Most importantly it tells us the amount of memory needed, the number of calculations required, and the amount of "filtering" that it can do. Basically, the more taps in a filter results in better stop band attenuation (less of the part we want filtered out), less ripple (less variations in the pass band), and steeper roll off (a shorter transition between the pass band and the stop band).

- Multiply-Accumulate (MAC) – In the context of FIR Filters, a "MAC" is the operation of multiplying a coefficient by the corresponding delayed data sample and accumulating the result. There is usually one MAC per tap.

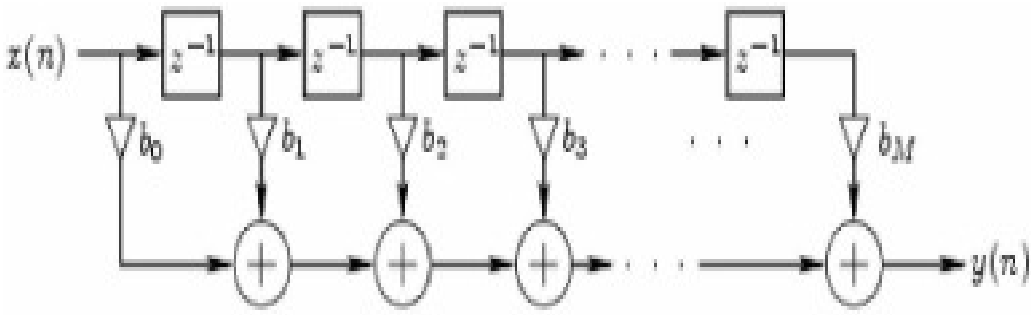


**FIGURE 11:** General causal FIR filter structure

Figure 11 gives the signal flow graph for a general finite-impulse-response filter (FIR). Such a filter is also called a transversal filter, or a tapped delay line. The implementation is one example of a direct-form implementation of a digital filter. The impulse response h(n) is obtained at the output when the input signal is the impulse signal □=[1 0 0 0…]. If the kth tap is denoted $b_k$, then it is obvious from figure 11 above that the impulse response signal is given by

$$h(n) = \begin{cases} 0, & n < 0 \\ b_n, & 0 \le n \le M \\ 0, & n > M \end{cases} \tag{19}$$

In other words, the impulse response simply consists of the tap coefficients, pretended and appended by zeros.

## 5. EXPERIMENTAL RESULTS AND CONCLUSION

### 5.1 Spectral Sharpening For Speech Enhancement In Matlab

Validation of the proposed filter was conducted using simulation tool. MATLAB v7.01 was used as platform. Speech signal constituting of 26000 samples at 8k samples/ sec was generated using wave recorder of windows and captured as a Matlab data file. This constitutes 3.25s of voice data. The waveform generated is presented in figure 12. The peak value of the signal generated is 275 mV. This signal formed the input to the hearing aid appliance. The output of a single stage is presented in figure 13 for $\beta$=0.04, $\gamma$=0.6 and $\mu$=0.98. From the observation of the filter output it is seen that the output amplitude is nearly 600 mV. The single stage output with the filter parameters, $\beta$=0.4, $\gamma$=0.6 and $\mu$=0.98 is presented in figure 14. In this case, peak amplitude is 390 mV which constitute a gain less than 2.

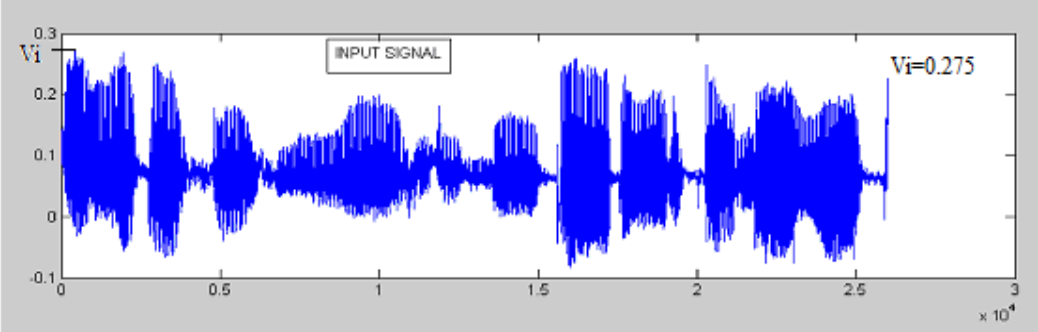


**FIGURE 12:** Waveform of the 3.25s speech input

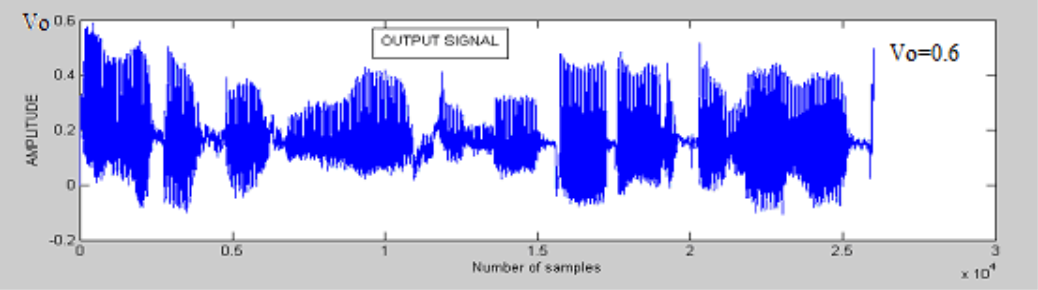


**FIGURE 12:** Waveform of the 3.25 second hearing aid output using parameters $\beta$=0.04,$\gamma$=0.6,$\mu$=0.98

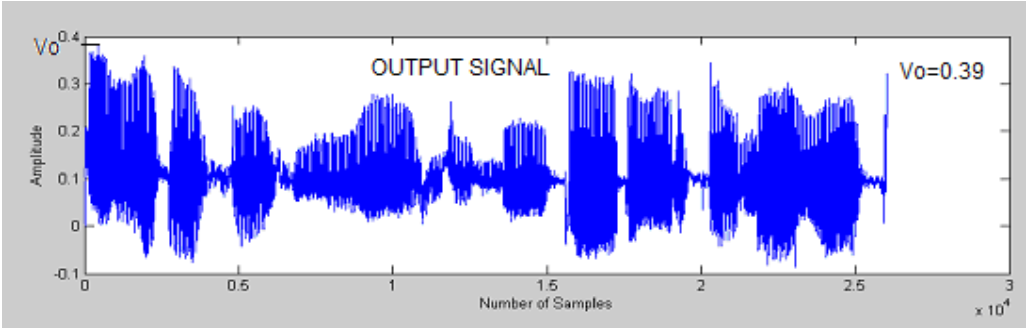


**FIGURE 13:** Waveform of the 3.25 second hearing aid output using parameters □=0.4,□=0.6,□=0.98

Following this; the performance of a 8 stage filter is observed. The filter output for β=0.04, γ=0.6, μ=0.98 and β=0.4, γ=0.6 and μ=0.98 are presented in figure 15 and figure 16 respectively. From figure 12 and figure 15, it is seen that output is more than double. Considering the superior performance of the 8 stage filter output over single stage filter, a 8 stage is used for hardware implementation.

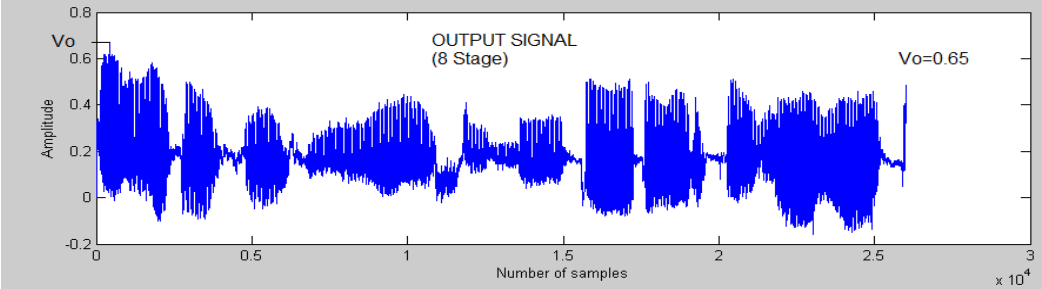


**FIGURE 15:** Waveform of the 3.25 second hearing aid output using parameters

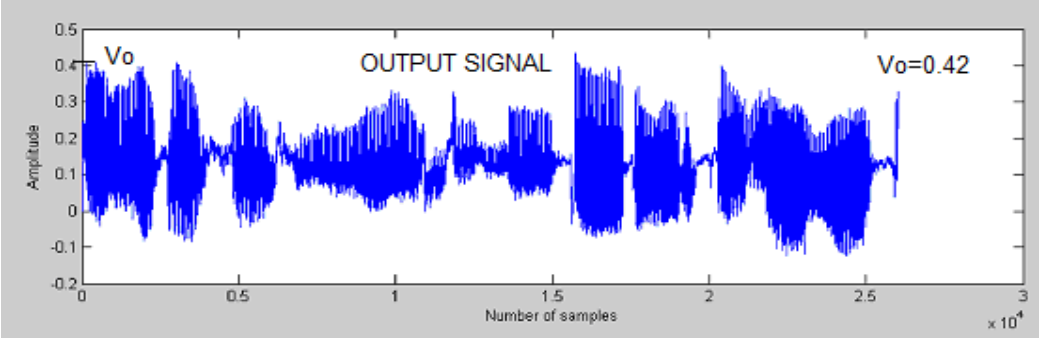


**FIGURE 16:** Waveform of the 3.25 second hearing aid output using parameters □=0.4,□=0.6,□=0.98

### 5.2 FPGA Based Simulation Results.

**MULTIPLIERS**

The table below compares the cell usage of the three multipliers (SHIFT/ADD, BOOTH'S and BOOTH WALLACE multiplier) for 8 bit by 8 bit multiplication and 16 bit by 16 bit multiplication. From the table we can see that the booth Wallace multiplier uses less hardware compared to that of the shift/add multiplier and booth multiplier. The details are given table 2.

| Cell Usage | Shift/add multiplier (8x8) | Shift/add multiplier (16x16) | Booth multiplier (8x8) | Booth multiplier (16x16) | Booth Wallace multiplier (8x8) | Booth Wallace multiplier (16x16) |
|---|---|---|---|---|---|---|
| BELS | 240 | 1000 | 333 | 975 | 167 | 697 |
| LUT-1 | 1 | 1 | 0 | 0 | 0 | 0 |
| LUT-2 | 14 | 1 | 37 | 36 | 5 | 9 |
| LUT-2 | 34 | 186 | 28 | 66 | 51 | 234 |
| LUT-4 | 74 | 290 | 116 | 399 | 83 | 328 |
| MUXCY | 56 | 240 | 64 | 228 | 0 | 0 |
| MUXF5 | 11 | 27 | 14 | 2 | 28 | 126 |
| XORCY | 49 | 225 | 61 | 219 | 0 | 0 |

**TABLE 2:** Cell used for the three Multipliers in virtex2p

| Cells used | Slices | 4-LUT | IO | Delay | Power consumption |
|---|---|---|---|---|---|
| Booth multiplier | 109 | 192 | 32 | 24.41 ns | 5 mW |
| Booth Wallace multiplier | 76 | 139 | 32 | 20.62 ns | 3 mW |

**TABLE 3:** Power consumption and Delay for two multipliers with 8x8 bits

From the table 3 we can see that the using the booth Wallace multiplier consumes less power compared to the booth multiplier and also that booth Wallace multiplier is faster than booth multiplier. Hence the Booth Wallace multiplier [17-20] is used for hearing aid design in VHDL in this investigation. The table 4 gives area used by the two multipliers. From this it can be seen that the booth Wallace tree multiplier uses less hardware than other.

| Cells used | Slices | Slice flip flops | 4-LUT | Logic | Shift registers | IO |
|---|---|---|---|---|---|---|
| Booth multiplier | 2684 | 183 | 5003 | 4979 | 24 | 32 |
| Booth Wallace Multiplier | 2583 | 196 | 4885 | 4866 | 19 | 32 |

**TABLE 4**: Cell usage for hearing aid component in virtex2p

### 5.3 Spectral Sharpening for Speech Enhancement in VHDL

The amplitude values of the speech signal sampled at 8 kS/s is rounded to 8 bits and stored in a text file for VHDL simulation. The hearing aid is designed in VHDL and is tested using different multipliers. The first 250 samples are taken as input for the hearing aid in VHDL .The output obtained through simulation is stored in a text file. The text file is read in MATLAB and is plotted as shown in the figure 18. The parameters used in VHDL are $\beta$=0.04, $\gamma$=0.6, $\mu$=0.98.

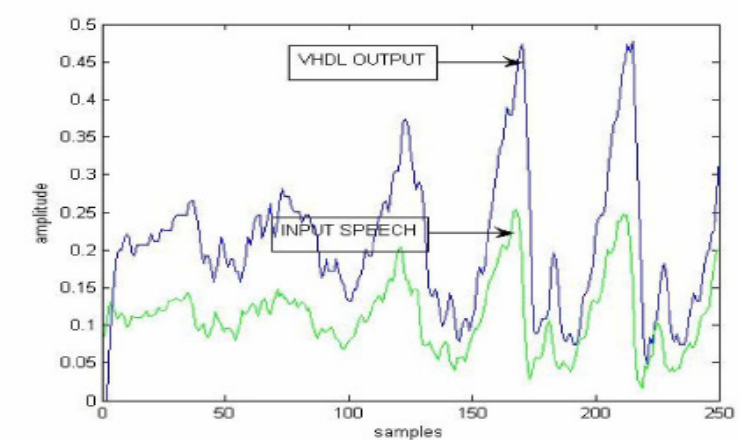


**FIGURE 14:** Comparision of input speech signal with output using vhdl for 250 samples

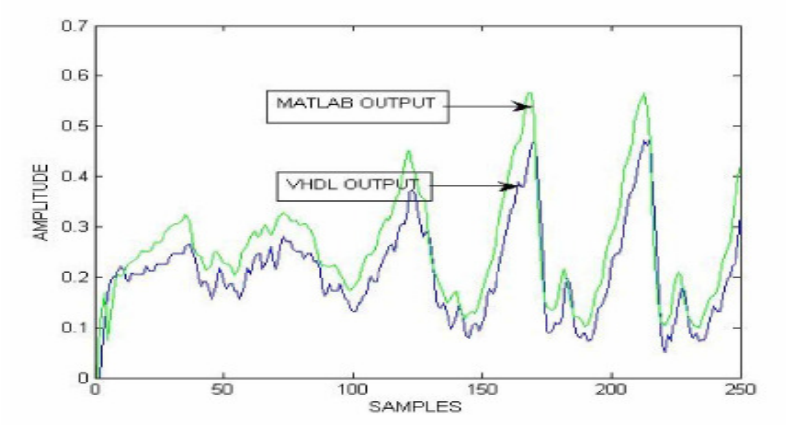


**FIGURE 15:** Comparison of the MATLAB output speech signal with the output obtained using VHDL for 250 samples

From the figure we can see that the output obtained using VHDL is slightly less in magnitude than MATLAB output. This is due to rounding of the values and due to fixed point multiplication. But from the figure 19 we see that the VHDL output follows the MATLAB output. The table 5 below shows the resource utilization summary and power consumed by the two designs using different multipliers.

| Design name | Slice used out of 3008 | 4-LUTs used out of 6016 | Slice FFs used out of 6016 | Shift registers | Logics | IO used out of 348 | Power consumption |
|---|---|---|---|---|---|---|---|
| High pass filter using Booth multiplier | 241 (7%) | 446 (7.4%) | 41 (<1%) | - | - | 25 (7%) | - |
| Synthesis filter | 175 (5.8%) | 323 (5.3%) | 32 (<1%) | - | - | 41 (11%) | - |
| Decorrelator | 164 (5.4%) | 292 (4.9%) | 104 (1.7%) | - | - | 40 (11%) | - |
| Hearing aid using Booth multiplier | 2684 (89%) | 5003 (83%) | 183 (3%) | 24 | 4979 | 32 (9.5%) | 40 mW |
| Hearing aid using Booth Wallace multiplier | 2583 (85%) | 4885 (81%) | 196 (3.3%) | 19 | 4866 | 33 (9.5%) | 30 mW |

**TABLE 5:** FPGA resources used and power of hearing aid design

## 6. Conclusions

All the papers referred so far do not convey any information about the power consumption and only architectural part is discussed. Our emphasis is to design an adaptive algorithm based on Booth-Wallace tree multiplier which consumes less power with respect to the use of Booth multiplier suitable for hearing aid application.. with this effort we the whole system in figure 10 is implemented using Booth-Wallace tree multiplier and power calculation of the whole system is done using Xilinx XPower Analyser. From the figure 19 it can be seen that our design output matches the Matlab output. Also referring to table 5 we can see that the power consumed by the hearing aid with booth Wallace tree multiplier is less than the hearing aid using booth multiplier which is about 25% lesser than the latter. So we can conclude that the hearing aid using booth Wallace tree multiplier consumes less power in this case.

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# Multi user detection in CDMA system using Linear and Nonlinear Detector

**Smita Parija** smita.parija@gmail.com
*Asst.Prof/ECE Deptt, Synergy Institute of Technology. Bhubaneswar, Orissa, India, 752101*

**Sandigdha Acharya** sandigdhaacharya@yahoo.co.in
*Lecturer/CSE Deptt, Synergy Institute of Technology Bhubaneswar,Orissa, India,752101*

**Abstract**

DS-Code division multiple access is considered as the third generation of cellular mobile used in interim standard 95(IS-95) [1]and it is currently being standardized for universal mobile telecommunication systems (UMTS). CDMA offers attractive features, such as frequency reuse, soft handoff, increased capacity, and multipath combating. In a CDMA system, several users simultaneously transmit information over a common channel using pre-assigned codes. The conventional single user detector consists of a bank of filters matched to the spreading codes. This detector suffers from two problems. First, multiple access interference (MAI) produced by the other co-channel users is a significant limitation to the capacity of this detector. The second problem is the near-far effect which occurs when the relative received power of interfering signals becomes larger. A potential solution is multi-user detection which exploits the information of signals of interfering users. In the present study performance of various linear detectors like matched filter detector, MMSE detector, and adaptive LMS detector are studied. These are the linear detectors that operate linearly on the received signal statistics and are suboptimal detectors. The matched filter bank is the conventional detector and offers the simplest way of demodulating CDMA signals .The detector resulting from the MMSE (minimum mean square error) criterion shows better performance over the conventional one for low SNR value. Adaptive LMS is employed to enhance the BER performance in MUD application.Several factors motivated the research to apply neural network as multi-user detector. NN are nonlinear classifier in addition to being adaptive and computationally efficient. The performance of two layer perceptron neural network using BP learning rule is used for multi-user detection of CDMA signals in AWGN channels. The neural network detectors show improvement of BER in the comparative analysis done in the present work. and offers further research scope for solving multi-user detection problems in CDMA application.

**Keywords:** MAI, CDMA, MMSE, LMS, NN Detector.

## 1. INTRODUCTION

The Design and implementation of a high-speed, high-quality, wireless link between two mobile terminals, located anywhere in the world is the challenge being faced by the communications research community today. The dramatic rise of the demand for the wireless mobile communications services over the recent years has emphasized the importance of efficient use of frequency bandwidth. Since the bandwidth available for mobile services is limited, various multiple access techniques have been proposed to increase the channel capacity, i.e. the number of users that can be supported within a specific geographical area. Traditionally, these techniques are based on frequency, time and code allocation.

The technique based on the division of the available spectrum into frequency bands which are then assigned to mobile users is Frequency Division Multiple Access (FDMA). FDMA is used in the first generation analogue systems. The second generation cellular mobile systems, such as the European GSM standard and the USA's Interim Standard IS-54 [6] have one common feature – they use Time Division Multiple Access (TDMA) to enable simultaneous access of mobile users. Unlike FDMA, in a TDMA system each user accesses the whole of the assigned bandwidth, but only for a fraction of time and on a periodic basis.

Code Division Multiple Access (CDMA) is used in Interim Standard 95 and it is currently being standardized for Universal Mobile telecommunications System (UMTS)]. The CDMA technique assigns uncorrelated codes to the mobile users, thus enabling them to access the full bandwidth, and for the complete duration of the call. This feature gives CDMA the advantage over FDMA and TDMA schemes.

CDMA (Direct Sequence Code Division Multiple Access) is considered as the third generation of cellular mobile, indoor wireless and personal communication systems. CDMA offers attractive features, such as frequency reuse, soft handoff, increased capacity and multipath combating.

In a CDMA system, a communication channel with a given bandwidth is accessed by all the users simultaneously. The different mobile users are distinguished at the base station receiver by the unique spreading code assigned to the users to modulate their signals. Hence, the CDMA signal transmitted by any given user consists of that user's data which modulates the unique spreading code assigned to that user which in turn modulates a carrier (the frequency of which is the same for all users), using any well-known modulation scheme such as binary phase shift keying (BPSK). Figure 1 shows the modulation of the bits of the users by a spreading code[4].

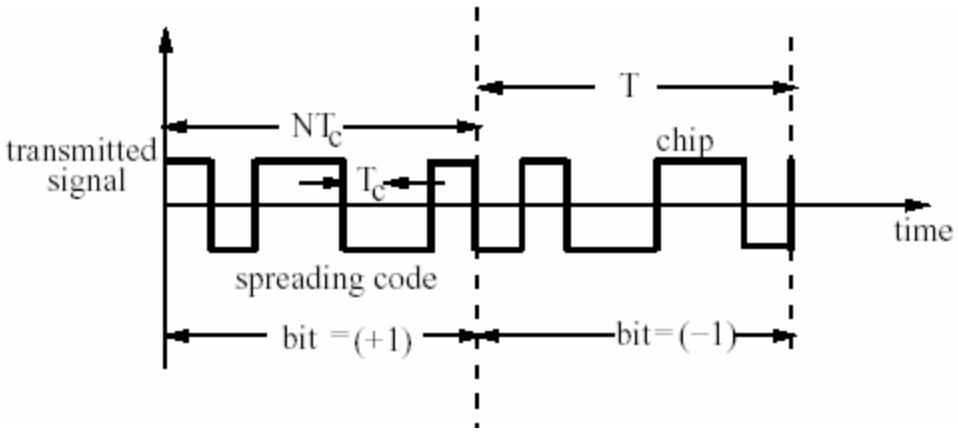


**FIGURE 1:**Spreading in a direct sequence CDMA system. The transmitted signal consists of 2 bits +1 and -1. Each bit is multiplied by a spreading code f+1,-1,+1,+1,- 1,-1,+1g consisting of 7 chips. T is the bit period, Tc is the chip period, and N is the number of chips per bit.

The low cross-correlation between the spreading codes of various users and peaky auto-correlation property of each code provide the basis for detection of the transmitted symbols of each user at the receiver. Wireless systems involve two radio links: the reverse link or the uplink from the mobile to the base station, and the forward link or the downlink from the base station to the mobile. Gold code generators are used extensively in Code Division Multiple Access . The Gold code generators use efficiently implemented Linear Feedback Shift Registers In a multi-user CDMA system several forms of "Spread Spectrum" modulation techniques are used. The most popular is the Direct Sequence Spread Spectrum (DS-SS). In this form of modulation each user signal is uniquely coded and spread across a wide band of transmission frequencies. Pseudo-random Noise (PN) sequences that are orthogonal to each other are used to code the user signals. Two sequences are considered orthogonal when their cross correlation coefficient is zero[4].

The first aim is to calculate the bit error rate of the linear detector like matched filter bank, MMSE detector, LMS detector. Then to determine the SNR for non linear detector using the neural network. Here multilayer perceptron is used by using the back propagation algorithm. It shows the better bit error rate performance for the nonlinear detector than the linear one but it has been seen when the no of user's increases linear detector shows the poor performance.

In a CDMA system, several users simultaneously transmit information over a common channel using preassigned codes. The conventional single user detector consists of a bank of filters matched to the spreading codes and then deciding on the sign of the outputs. This detector suffers from two problems. First, Multiple Access Interference (MAI) produced by the other co-channel users is a significant limitation to the capacity of this detector. The second problem is the near-far effect, which occurs when the relative received power of interfering signals becomes larger.

A potential solution is multi-user detection ,which exploits the information of the signals of interfering users. The optimum multi-user detector evaluates a log-likelihood function over the set of all possible information sequences. It achieves low error probability at the expense of high computational complexity, which increases exponentially with the number of users. So this method is extremely complex for a realistic number of users. Consequently, there has been considerable research into suboptimal detectors. These detectors achieve significant performance gains over the conventional detector without the exponential increase in receiver complexity. Several factors motivate us to apply Neural Networks (NN) as multi-user detectors[11]. They are adaptive and computationally efficient. Also, the cyclostationary structure of MAI and nonlinear decision boundaries formed by an optimal receiver in CDMA can be estimated by NN Aazhang et al. first reported a study of a multilayer perceptron NN in CDMA systems, and showed that in the case of applying a complicated algorithm named assisted BP, in which the number of hidden layer nodes grows exponentially with the number of users, its performance is close to that of the optimum receiver in both synchronous and asynchronous Gaussian channels.

## 2. THE SYSTEM MODEL OF CDMA :

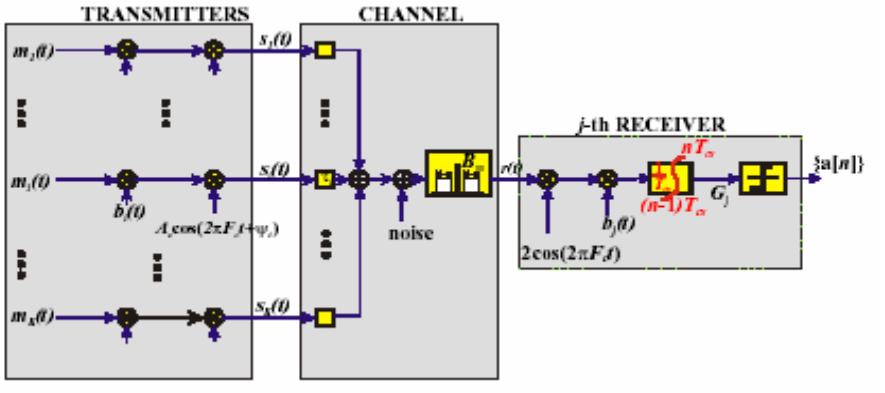


**FIGURE 2:** TRANSMITTER MODEL

The system model consists of K independent simultaneous users. The kth user's transmitted signal assuming BPSK data modulation is of the form.

$$Y_k(t) = \sum_k \sqrt{E_k(i)} b_k(i) s_k(t - iT) \quad (1)$$

Where $E_k(i)$ is the power of the kth user at time iT,1/T is the data rate, $b_k \varepsilon \{\pm 1\}$ is the data bit of user k during the ith interval, and $s_k$ (t) is the spreading (signature) waveform of duration T and normalized power which is composed of a spreading sequence of N chips (code length) as

$$s_k(t) = \sum_{n=0}^{N-1} a^k{}_n(t)\, p(t - nT_c) \quad (2)$$

Where $a^k{}_n \varepsilon$ (-1, 1) is the spreading sequence, p(t) is the rectangular waveform of duration $T_c$, and T = $nT_c$. We obtain the receiver input and output in AWGN and fading Channels.

## 3. MULTIUSER-DETECTION

Multiuser detection is a technology that spawned in the early 80's. It has now developed into an important, full-fledged field in multi-access communications. Multiuser Detection (MUD) is the intelligent estimation/demodulation of transmitted bits in the presence of Multiple Access Interference (MAI). MAI occurs in multi-access communication systems (CDMA/ TDMA/ FDMA) where simultaneously occurring digital streams of information interfere with each other. Conventional detectors based on the matched filter just treat the MAI as additive white gaussian noise (AWGN).However, unlike AWGN, MAI has a nice correlative structure that is quantified by the cross-correlation matrix of the signature sequences. Hence, detectors that take into account this correlation would perform better than the conventional matched filter-bank. MUD is basically the design of signal processing algorithms that run in the black box shown in figure These algorithms take into account the correlative structure of the MAI.

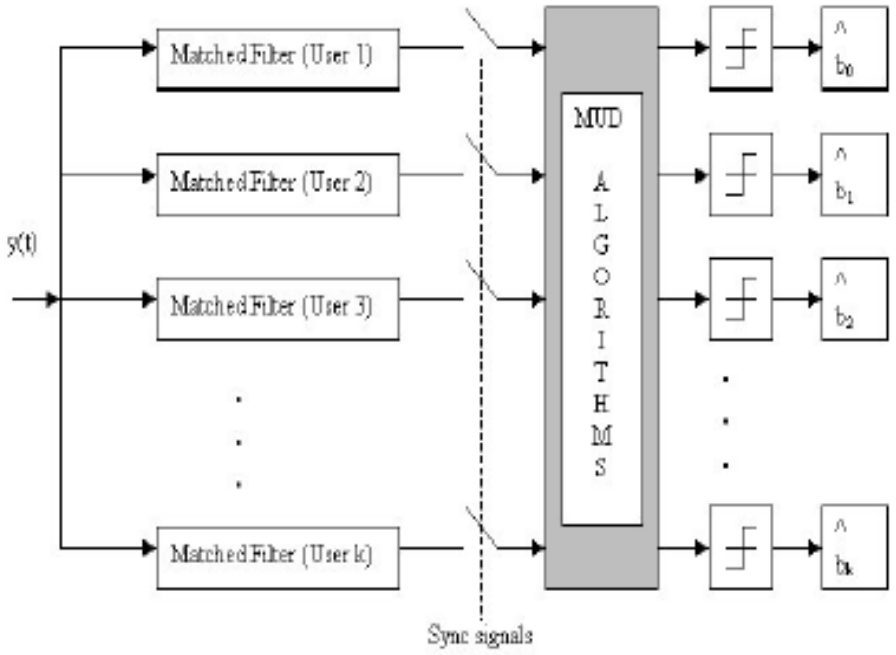


**FIGURE 3.1:** A matched filter bank

The decision statistic a the output of the Kth matched filter is given by

$$y_k = \int_0^T y(t)s_k(t)dt \tag{3}$$

where y(t) and sk(t) is given by (1) and (2). Expanding the above equation

$$y_k = \int_0^T \{\sum_{j=1}^{K} A_j b_j s_j(t) + n(t)\} s_k(t)dt \tag{4}$$

Using eq(3)

$$y_k = \sum_{j=1}^{k} A_j b_j \rho_{jk} + n_k \tag{5}$$

$$n_k = \int_0^T n(t)s_k(t)dt \tag{6}$$

Where $\rho_{11} = 1$ and y simplifies to

$$y_k = A_k b_k + \sum_{j=1, j\neq k}^{K} A_j B_j \rho_{jk} + n_k \tag{7}$$

The 2nd term in the above eq is the MAI. The matched filter treats the MAI just as white noise. The noise variance at the output of the matched filter is given by

$$E(n_k{}^2) = E[\int_0^T n(t)s_k(t)dt \int_0^T n(s)s_k(s)ds] = \int_0^T \int_0^T E[n(t)n(s)]s_k(s)s_k(t)dtds$$
$$= \int_0^T \int_0^T N_o\delta(t-s)s_k(t)dtds = \int_0^T N_o s_k{}^2(t)dt = N_o \quad (8)$$

Similarly, the noise covariance can be shown to be

$$E(n_i n_j) = N_0\rho_{ij} \quad (9)$$

Hence the noise covariance matrix can be defined as

$$E[nn^T] = \{N_0\rho_{ij}\} = N_0 R \quad (10)$$

where R is given by (4) and $n = [n_1, n_2 ....., n_k]^T$ . Stacking up (2.5) for all the users we get

$$\begin{bmatrix} y_1 \\ y_2 \\ \vdots \\ y_k \end{bmatrix} = \begin{bmatrix} \rho_{11} & \rho_{12} & \cdots & \rho_{1k} \\ \rho_{21} & \rho_{22} & \cdots & \rho_{21k} \\ \vdots & \vdots & \cdots & \vdots \\ \rho_{k1} & \rho_{k2} & \cdots & \rho_{kk} \end{bmatrix} \begin{bmatrix} A_1 & 0 & \cdots & 0 \\ 0 & A_2 & \cdots & 0 \\ \vdots & \vdots & \cdots & \vdots \\ 0 & 0 & \cdots & A_k \end{bmatrix} \begin{bmatrix} b_1 \\ b_2 \\ \vdots \\ b_k \end{bmatrix} + \begin{bmatrix} n_1 \\ n_2 \\ \vdots \\ n_k \end{bmatrix} \quad (11)$$

In matrix notation we have,

$$y = RAb + n \quad (12)$$

It is observed that as the MAI increases (the number of users increases) the performance becomes poor. This is because the detector ignores the cross-talk between users (the MAI) as white noise. Good MUDs, as described in the next few sections, take into the account the correlative property of the cross-talk.

### 3.1 Limitations of the conventional detector

Although {y1,y2,…,$y_k$} are sufficient statistics for detecting {b1,b2,…,$b_k$}, $y_k$ is not a sufficient statistic for detecting bk. The conventional detector makes the mistake of making this assumption($y_k$ is a sufficient statistic for detecting bk) by ignoring the MAI as background noise. This is one reason for the poor performance of the matched filter bank when the number of users are large. Another serious limitation of the conventional detector is that it is seriously affected by the near-far problem.This causes a significance degradation in the system performance even when the number of users is very small. Adapting (3.9) to the 2 user scenario we get the fact that Q is a monotonically decreasing function was used to get the upper bound. If the interferer is not dominant , the bit error probability is less than half. But if the interferer is dominant (near-far problem) the bound becomes greater than half. Consider the case when there is no noise in the system and the interferer is dominant, Here we see that in the absence of noise, though highly hypothetical, the matched filter receiver reduces to flipping a coin and deciding the output bits. This is an undesirable feature of the conventional detector (may perform better in the presence of noise than in the absence of noise).

### 3.2 The MMSE Linear Detector

At low SNRs, the matched filter bank performs better than the decor relating detector as observed from figure 3.6. Hence, it might be possible to improve the performance by incorporating some SNR information in the MUD algorithms. In this section, one such approach is investigated where the mean squared error between the output and data is minimized. The detector resulting from

the MMSE (minimum mean square error) criterion is a linear detector[1]. Two different adaptive approaches of the MMSE linear detector are also studied at the end of this section. One of the approaches requires no prior information of the SNRs or the signature waveforms but requires a training sequence to adapt and compute the optimum weights to be applied on the received statistic. The other approach does not need a training sequence but requires exact knowledge of the signature sequence. Being a linear detector like the decor relating detector, the MMSE receiver also weights thereceived statistic y with a weight vector w to form the decision statistic[1]. It has been proved that minimizing the MSE at the output of the linear transformation is equivalent to maximizing the SIR at the output of the linear transformation. The optimal value of the minimizes the MSE between the weighted received statistic and the transmitted bit is derived in the next section. ..The receiver structure for user m is
shown in figure.

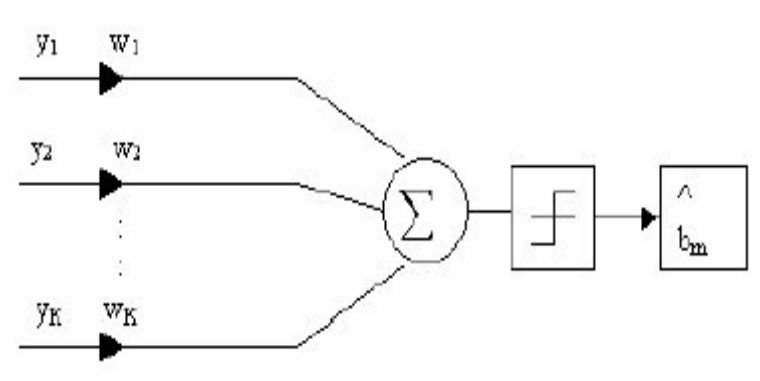


**FIGURE 3.2**: MMSE linear transformation for user m.

### 3.2.1 Optimal Weights for an MMSE Linear Detector in an AWGN Channel

The MMSE linear detector for user 1 determines a waveform c1(t) such that the MSE error between the transmitted bit and the correlation between c1(t) and the received signal y(t) is minimized. The objective function (the mean square error in this case) is defined as

$$\psi(c1) = E\left\{\left(b_1 - \langle c1, y\rangle\right)^2\right\} \tag{13}$$

In the finite dimensional representation of the above eq can be expressed as

$$\psi(w_1, \quad w_2, \quad ..., \quad w_K) = E\left\{\left(b_1 - \sum_{i=1}^{K} w_i y_i\right)^2\right\} \tag{14}$$

Where $\{w_1, w_2, \ldots, w_k\}$ are the weights operating on the received statistic
$\{y_1, y_2, \ldots, y_k\}$. Representing the above eq in a compact and convenient matrix notation,

$$\psi(w) = E\left\{\left(b_1 - w^T y\right)^2\right\}$$

Using linearity of the Expectation operator,

$$\psi(w) = E(b_1^{\,2}) - E(2b_1 w^T y) + E\left\{\left(w^T y\right)\left(w^T y\right)^T\right\}$$

$$\psi(w) = 1 - 2w^T E(b_1 y) + E\left\{w^T y y^T w\right\} \tag{15}$$

Since the bits of user $1 E(b_1^{\,2})=1$, Therefore,

$$\psi(w) = 1 - 2w^T E(b_1 y) + w^T E\left\{yy^T\right\} w \tag{16}$$

From eq 15, we have

$$y = RAb + n$$

Consider, $E(b_1 y) = E(b_1 RAb + b_1 n)$

$$= RAE\left( b_1 \begin{bmatrix} b_1 \\ b_2 \\ \vdots \\ b_K \end{bmatrix} \right) + E(b_1 n)$$

$$= RA \begin{bmatrix} E(b_1^2) \\ E(b_1 b_2) \\ \vdots \\ E(b_1 b_K) \end{bmatrix} + b_1 E(n) \qquad (17)$$

Since the bits of user 1 are uncorrelated with the bits of other users we have,

$$E(b_1 b_K) = \begin{cases} 0 & , i \neq j \\ 1 & , i = j \end{cases} \qquad (18)$$

Using eq 17 and the fact that the noise n is zero mean i.e., E(n)=0 in 3.26

$$E(b_1 y) = RA\begin{bmatrix} 1 & 0 & \cdots & 0 \end{bmatrix}^T \qquad (19)$$

Using the definition of A and R

$$E(b_1 y) = \begin{bmatrix} \rho_{11} & \rho_{12} & \cdots & \rho_{1K} \\ \rho_{21} & \rho_{22} & \cdots & \rho_{2K} \\ \vdots & \vdots & \cdots & \vdots \\ \rho_{K1} & \rho_{K2} & \cdots & \rho_{KK} \end{bmatrix} \begin{bmatrix} A_1 & 0 & \cdots & 0 \\ 0 & A_2 & \cdots & 0 \\ \vdots & \vdots & \cdots & \vdots \\ 0 & 0 & \cdots & A_K \end{bmatrix} \begin{bmatrix} 1 \\ 0 \\ \vdots \\ 0 \end{bmatrix}$$

$$\therefore E(b_1 y) = \begin{bmatrix} \rho_{11} A_1 \\ \rho_{21} A_1 \\ \vdots \\ \rho_{K1} A_1 \end{bmatrix} \qquad (20)$$

Now consider the second expectation term in eq 3.22

$$E\{yy^T\} = E\{(RAb)(RAb)^T\} + E(nn^T)$$
$$= E\{RAbb^T A^T R^T\} + N_o R \qquad (21)$$

Using the fact that A and R are symmetric matrices, we get

$$E\{yy^T\} = RAE\{bb^T\}AR + N_o R$$
$$= RA^2 R + N_o R \qquad (22)$$

Substituting eq 20and eq 22 in eq 15

$$\psi(w) = 1 - 2w^T \begin{bmatrix} \rho_{11}A_1 & \rho_{21}A_1 & \cdots & \rho_{K1}A \end{bmatrix}^T + w^T (RA^2R + N_oR)w \quad (23)$$

The above equation gives the objective function (MSE) that should be minimized according to the MMSE criterion. Performing a matrix derivative operation on (14) we get,

$$W_{opt} = (R + N_0A^{-2})^{-1} \quad (24)$$

Where $$N_0A^{-2} = diag\left\{ \frac{N_0}{A_1^2}, \quad \frac{N_0}{A_2^2}, \quad \cdots, \quad \frac{N_0}{A^2_k} \right\} \quad (25)$$

The MMSE detector requires the SNR information and hence again precomputation of the matrix inverse is not a feasible solution. Also, getting good estimates of the SNR is not temporally efficient. Therefore, it would be nice if there was some way to eliminate the need to compute matrix inverses and the need to have apriori information (signature sequences) and other additional information (SNR) for decoding. This objective can be realized through adaptive MUD algorithms. Adaptive algorithms "learn" the desired filter response from the received signals. There are different approaches to implement the "learning" capability.Two approaches will be studied in the next sub- calls for a training sequence.The second approach doesn't require any training sequence but requires exact knowledge of the signature sequences of the users and also takes longer to converge.

**3.3 system model(neural network)**

The optimum multi-user detector evaluates a log-likelihood function over the set of all possible information sequences. It achieves low error probability at the expense of high computational complexity that increases exponentially with the rule for multi-user detection of DS/CDMA[7-9] signals in AWGN(Additive White Gaussian Noise) and multipath fading channels The results show superior improvement over the previous studies in terms of the receiver complexity .

Therefore, this method is extremely complex for a realistic number of users. Consequently, there has been considerable research into suboptimal detectors. These detectors achieve significant performance gains over the conventional detector without the exponential increase in the receiver complexity. In this section, we explain multilayer perceptron and Hopfield neural networks. We first describe the back propagation (BP) algorithm for training multilayer perceptron. Since our goal is to improve the performance of BP neural network, subsequently we explain different training algorithms and criterion that have shown better performance than the BP in radar, sonar, speech, and pattern recognition applications. Then Hopfield neural network is explained.

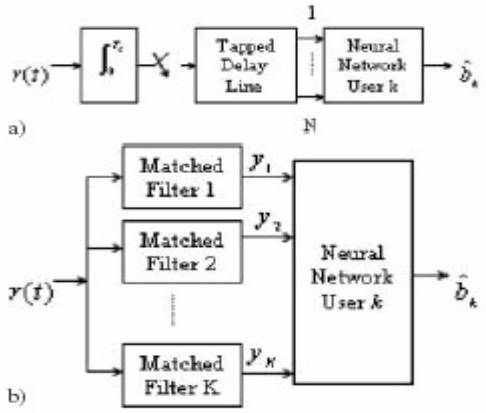


**FIGURE 3.2**: Two kinds of processing of the received signal.

In this section, we explain multilayer perceptron and Hopfield neural networks. We first describe the back propagation (BP) algorithm for training multilayer perceptron.

Since our goal is to improve the performance of BP neural network, subsequently we explain different training algorithms and criterion that have shown better performance than the BP in radar, sonar, speech, and pattern recognition applications. Then Hopfield neural network is explained. In this section, we explain multilayer perceptron and Hopfield neural networks. We first describe the back propagation (BP) algorithm for training multilayer perceptron. Since our goal is to improve the performance of BP neural network, subsequently we explain different training algorithms and criterion that have shown better performance than the BP in radar, sonar, speech, and pattern recognition applications. Then Hopfield neural network is explained.

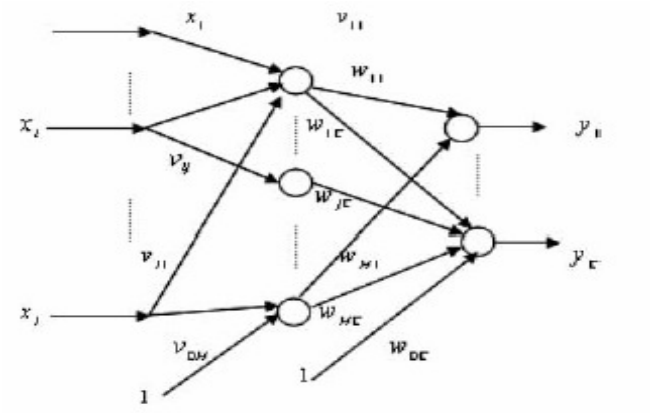

**FIGURE 3.3:** The structure of a typical two-layer perceptron neural network

Multilayer perceptron is a feed forward network where the outputs of each layer are applied to the inputs of the next layer. Figure 1 shows the structure of a typical
two-layer perceptron neural network containing one hidden layer and output layer. The parameters of network are defined as:

- The numbers of nodes in the input, hidden, and output layers are I, H, and C, respectively.
  - $x_i$: the ith input unit.
  - $v_{ij:}$ weight between the ith input unit and the jth unit of hidden layer.
  - $v_{0j}$ : bias weight;
  - $w_{jk}$: weight between the jth unit of hidden layer and the kth output;
  - $w_{ok}$: bias weight;
  - $z_{inj}$: the jth input unit of hidden layer;
  - $z_k$: thejth output of hidden layer;
  - $y_{ink}$: thekth input of output layer;

  $y_k$: the kth unit of output

$$zin_j = \sum_{i=1}^{l} x_i v_{ij} + v_{oj}$$ layer;

$$z_j = f(zin_j)$$

$$yin_k = \sum_{j=1}^{H} z_j w_{jk} + w_{ok}$$

$$y_k = f(yin_k)$$

- f(.): activation function;
- $t_k$: the desired output or target.

**3.4 Minimum mean square error (back propagation) Criterion**

In this common criterion, the objective of network training is to find the optimal weights to minimize the sum of square error between the desired outputs (targets) and actual outputs of net .

$$E = \frac{1}{2M}\sum_{m=1}^{M}\sum_{i=1}^{C}\left[t_i(m) - y_i(m)\right]^2 \tag{26}$$

Where M is the number of training patterns, C is the number of outputs, ti(m) is the ith component of the mth target (±1 in CDMA), and yi(m) is the ith output of the network for the mth input pattern. The weight updating is obtained according to the following rule:

$$W(new) = W(old) - \mu\frac{\partial E}{\partial W} \tag{27}$$

where W is the weights of the net (containing v and w) and μ is the learning rate. The weight change rules are as follows

$$\begin{cases} \Delta w_{jk} = \mu\delta_k z_j \\ \Delta w_{0k} = \mu\delta_k \quad \text{where} \quad \delta_j = \delta in_j f(zin_j) \end{cases} \tag{28}$$

$$\begin{cases} \delta v_{ij} = \mu\delta_k z_j \\ \delta v_{0j} = \mu\delta_j \quad \text{where} \quad \delta_j = \delta in_j f'(zin_j) \\ \delta in_j = \sum_{k=1}^{C}\delta_k w_{jk} \end{cases} \tag{29}$$

In CDMA application, we use bipolar sigmoid as activation function:

$$f(u) = \frac{1-e^{-u}}{1+e^{-u}} \Rightarrow f'(u) = (1-f(u))(1+f(u)) \tag{30}$$

Depending on the sign of the output of the network, the received signal will be classified to ±1. This network is approximation-based formulation net, i.e., aim is how close the result is to the expected value. In CDMA application, our goal is the classification of the received data, therefore it is only necessary to know the correctness of the classification. Hence we use decision based networks[12,13].

The first step is to feed the input vector through the network and compute every unit in the network. Recall that this is done by computing the weighting sum coming into the unit and then applying the sigmoid function. The second step is to compute the squared error of the network. Recall that this is done by taking the sum of the squared error of every unit in the output layer. The target vector involved is associated with the training sample (the input vector). The third step is to calculate the error term of each output unit, indicated below as 'delta'. The error term is related to the partial derivative of each weight with respect to the network error. The fourth step is to calculate the error term of each of the hidden units. The hidden unit error term depends on the error terms calculated for the output units. The fifth step is to compute the weight deltas. 'Eta' here is the learning rate. A low learning rate can ensure more stable convergence. A high learning rate can speed up convergence in some cases. The final step is to add the weight deltas to each of the weights. I prefer adjusting the weights one layer at a time. This method involves recomputing the network error before the next weight layer error terms are computed[8,10].

## 4. SIMULATION AND RESULT

Finally there is the simulation first done on the various nonlinear detector like matched filter bank, which is the conventional one consists of bank of filters. the simulation is done in order to get the better performance over the linear one like multilayer perceptron. Conventional detectors based on the matched filter just treat the MAI as Additive White Gaussian Noise (AWGN).Unlike MAI has a nice correlative structure that is quantified by the cross-correlation matrix of the signature sequences. Linear MUDs are detectors that operate linearly on the received signal

statistics i.e they perform only linear transformations on the received statistics. Then analysis done on the MMSE detectors where the mean square error between the output and data is minimized. The detector resulting from the MMSE (Minimum Mean Square Error)criteria is a linear detector. It has been shown that minimizing the MSE at the output of the linear transformation is equivalent to maximizing the bit error rate the output of the linear transformation.

The first example of non linear detector is matched filter bank. This section introduces and analyses the matched filter bank detector which was the conventional and most simplest way of demodulating CDMA signals (or any other set of mutually interfering digital streams). In conventional single-user digital communication systems, the matched filter is used to generate sufficient statistics for signal detection. In the case of a multi-user system, the detector consists of a bank of matched filters (each matched to the signature waveforms of different users in the case of CDMA)[11,14]. This type of detector is referred to as the conventional detector in MUD literature.

It is observed that as the MAI increases (the number of users increases) the performance becomes poor. This is because the detector ignores the cross-talk between users (the MAI) as white noise. Serious limitation of the conventional detector is that it is Seriously affected by the near-far problem. This causes a significant degradation in the system performance even when the number of users is very small. It is observed that at low SNRs the matched filter performs better. Hence, the decorrelating detector is not an optimal.

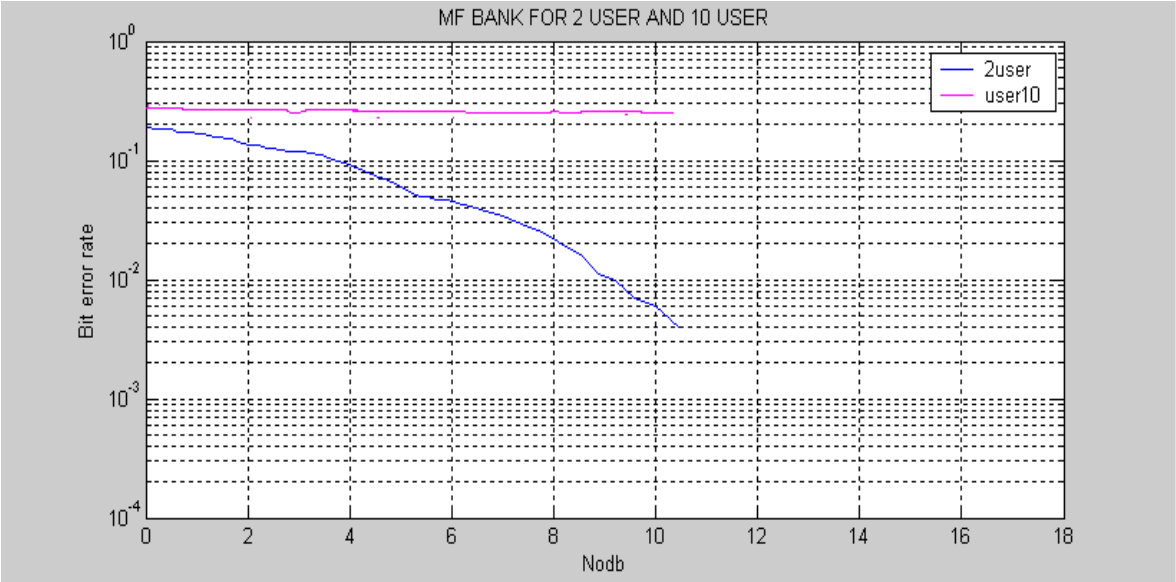


**FIGURE4.1:** Comparision of Matched filter bank of 2 user with 10 user

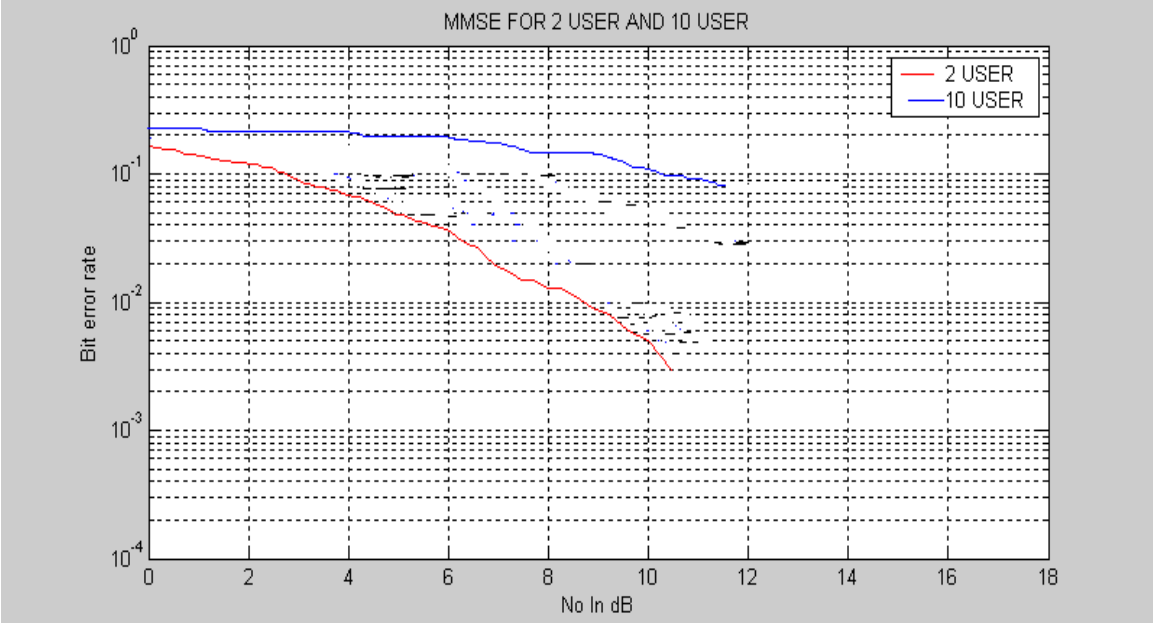


**FIGURE4.2:** Comparision of Matched filter bank of 2 user with 10 user

The second example of the linear detector is MMSE. At low SNRs, the matched filter bank performs better than the decorrelating detector as observed from figure. Hence, it might be possible to improve the performance by incorporating some SNR information in the MUD algorithms. In this section, one such approach is investigated where the mean squared error between the output and data is minimized. The detector resulting from the MMSE (minimum mean square error) criterion is a linear detector.

Two different adaptive approaches of the MMSE linear detector are also studied at the end of this section. One of the approaches requires no prior information of the SNRs or the signature waveforms but requires a training sequence to adapt and compute the optimum weights to be applied on the received statistic. The other approach does not need a training sequence but requires exact knowledge of the signature sequence. It has been proved that minimizing the MSE at the output of the linear transformation is equivalent to maximizing the SIR at the output of the linear transformation that the MMSE receiver maximizes the SIR at the output of the transformation shown in the above figure.

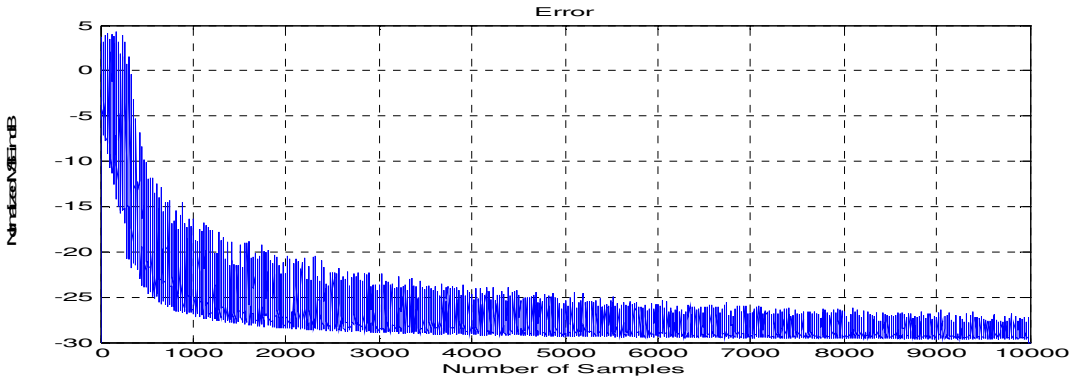


**FIGURE4.3:** Training curve for MLP for different samples .

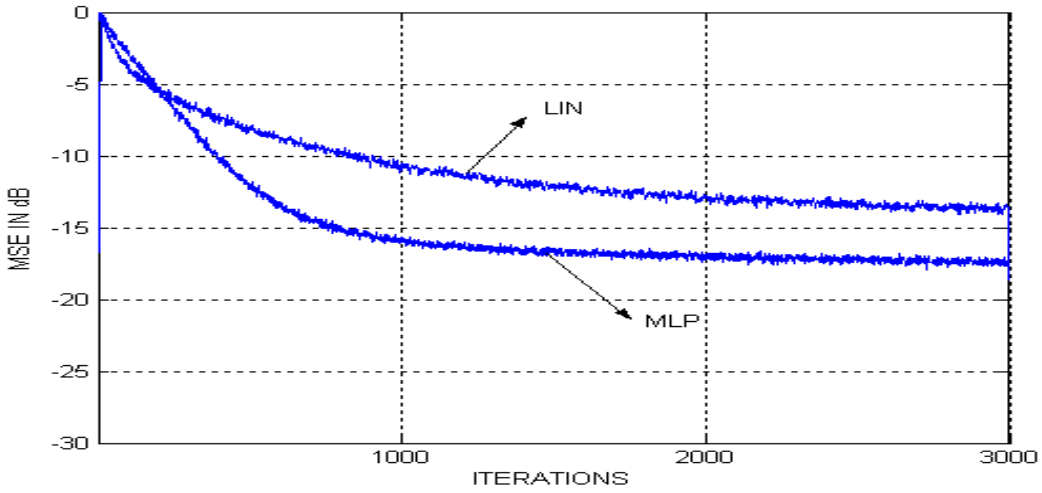


**FIGURE4.4:** Comparison of learning for linear and nonlinear detectors.

## 5. CONCLUSION AND FUTURE WORK

This thesis review gives a background on the fundamental concepts of linear and nonlinear detector. Different detectors like matched filter bank, MMSE detectors, and LMS detectors were studied. Further MLP based detector is suggested for CDMA detection which provides improvement in BER performance over the nonlinear one. For multi-user detection problem this Neural Network based detector[16] also has reduced structural configuration which helps for easier real time implementation. Faster learning using BP algorithm and with less no of training samples show that there is scope for its use in practical detectors.

Estimating the performance of linear and non linear detector has greater importance. Here the back propagation algorithm is proposed. Which provides better performance curve and training than the linear one. Instead of BP RLS can also be proposed which is having faster learning. Some aspects of the proposed algorithm are only briefly touched in this thesis and may be further investigated. The proposed algorithms are only for performance of ber which shows greater signal to noise ratio. The proposed algorithm for ber performance focused on only AWGN channel. It is proposed that the algorithm is to be further extended for fading channel signals since in fading channel shows better performance which is having greater importance as it involves multipath fading in it.

The performance of BP network[17,18] in AWGN channel with the conventional decorrelator multistage and optimum detectors widely used for comparative analysis. SVM (Support vector machine) also can be used as detectors. In fading channel the rake and single user lower bound receivers are considered for comparison. Since our goal is to improve the performance of BP net, we consider different neural networks.

We can apply decision based neural network (DBNN), fuzzy decision neural network (FDNN) discriminative learning, minimum classification. We also propose modified DBNN that outperforms DBNN. A comparison between BP perceptron and Hopfield neural nets can From the above results we can conclude that neural network can be used as multi-user detector in CDMA systems. Its performance depends on the parameters, where they are obtained by experiments. The number of training samples and hidden layer nodes and computational complexity increases with the number of users. The complexity of neural network is in the training phase that can be

organized in parallel. Of course the hardware implementation of neural network especially for large number of users in a realistic environment should be considered.

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# Optimization of Herbal Drugs using Soft Computing Approach

**S. K. Nayak** simanta.nayak@eastodissa.ac.in
*Asst. professor, Department of Computer Science & Engineering Eastern Academy of Science and Technology* Bhubaneswar, Orissar-754001, India

**Dr. P. K. Patra** hodcomputer@yahoo.co.in
*Prof. & Head, Department of Computer Science & Engineering* College of Engineering and Technology *Bhubaneswar, Orissar-751003, India*

**Dr. P. Padhi** payodharpadhi@gmail.com
*Dean, Research and Development Center Hi-Tech Medical College and Hospital, Health Park Bhubaneswar, Orissar-751025, India*

**A. Panda** aparajeya81@gmail.com
*Research Assistant, Research and Development Center Hi-Tech Medical College and Hospital, Health Park Bhubaneswar, Orissar-751025, India*

**Abstract**

The study presents the results of our investigation into the use of Genetic Algorithms (GA) and Artificial Neural Network (ANN) for identifying near optimal design parameters of compositions of drug systems that are based on soft computing approach for herbal drug design. Herbal medicine has been applied successfully in much clinical practices since long throughout in world. The present study proposes a novel concept using a computational technique to predict bioactivity of herbal drug and designing of new herbal drug for a particular disease. Genetic algorithm investigated the relationship between chemical composition of a widely used herbal medicine in India and its bioactivity effect. The predicted bioactivity with respect to its composition indicates that the proposed computing method is an efficient tool to herbal drug design.

**Keywords:** Herbal drugs, GA, ANN.

## 1. INTRODUCTION

In the last few years there has been an exponential growth in the field of herbal medicine and these drugs are gaining popularity both in developing and developed countries because of their natural origin and less side effects [1]. Many traditional medicines are derived from medicinal plants, minerals and organic matter [2]. A number of medicinal plants, traditionally used for over 1000 years named Rasayana are present in herbal preparations of Indian traditional health care systems [3]. In Indian systems of medicine, most practitioners formulate and dispense their own recipes [4]. Major hindrance in amalgamation of herbal medicine in modern medical practices is lack of scientific and clinical data proving their efficacy and safety. There is a need for conducting clinical research in herbal drugs, developing simple bioassays for biological standardization, pharmacological and toxicological evaluation, and developing various animal models for toxicity and safety evaluation [5]. It is also important to establish the active component/s from these plant extracts.

Drug discovery is a complex and costly process, which involves more time for making and testing New Composition Entities (NCE). The average cost of creating NCE in a pharmaceutical company was estimated about $ 1500/ compound [6]. Generally, herbal medicine is not composed of several herbs in appropriate proportion. The constituent of herbs and their proportion of certain herbal medicine are determined according to traditional medical knowledge. Unlike modern drugs in the form of single chemical ingredient, herbal medicine may contain hundreds of chemical compounds. Many researchers believe that synergistic effect of different active ingredients contributes to the therapeutic effect of herbal medicine [7]. Modern clinical trial has proved that herbal drug composed of multiple herbs with multiple compounds in certain proportion has greater efficacy than a single herb. Therefore, modern herbal drug can be produced as a combination of different active components from herbs. However, the complex chemical composition of herbal medicine leads to the lack of appropriate method for identifying active compounds and optimizing the formulation of herbal medicine. The variation of biologic activity of herbal medicine is tightly associated with the variation of their chemical composition. Such relationship between chemical composition and biological activity is regarded as Quantitative Composition–Activity Relationship (QCAR) [8]. By quantitatively analyzing the chemical composition and bioactivity relationship, mathematical model could be established to predict activity of herbal medicine. Moreover, an optimal combination of herbal medicine can be evaluated based on QCAR model, which enables us to integrate different active components to form a more effective herbal drug. In the present study, a soft computing approach has been proposed to model the composition–activity relationship method to predict the bioactivity of herbal drug in designing a new herbal drug for a specific disease.

## 2. MATERIALS AND METHODS

### 2.1 Herbal Drugs

Seven plants with reported antidiabetic property are taken for the study. The plants are Aloe Vera [9], Catharanthus roseus [10], Momordica charantia [11], Aegle marmelos [12], Aerva Lanata [13], Phyllanthus emblica [14] and Azadirachta indica [15]. The plant extracts are obtained by sox elation method. Different components are isolated by column chromatography and HPLC method. The biological activities of individual active component are studied in animal models.

### 2.2 Mathematical Model

QCAR is the correlation between chemical composition and biological activity of the drug system. Suppose 'D' is the herbal drug which is consists of 'n' number of herbs having total weight '$W_d$'. Each herbal contains maximum m number of components. Hence the herbal medicine D is a combination of different components, which represents by a row vector [$C_1$, $C_2$, $C_3$… $C_m$] n. Moreover each component is having separate biological activity, which can be denoted as a column vector says B = [$B_1$, $B_2$, …. $B_m$] $^T$. The whole matrix is a [n*m] matrix. If a single compositional bioactivity will change the overall bioactivity will change.
Hence B = f (C)

The function may be linear or nonlinear. But generally nonlinear relationship is more appropriate in case of herbal drug.

## 3. GENETIC ALGORITHMS AND ARTIFICIAL NEURAL NETWORK

The prediction of biological activity of a chemical compound from its compositional features, representing its physico-chemical properties, plays an important role in drug discovery, design and development. Since the biological data is highly non-linear, the machine-learning techniques have been widely used for modeling it. In the present work, genetic algorithm (GA) and artificial neural networks (ANN) are used to develop computational prediction models on a dataset of antidiabetic compounds. The hybrid GA-ANN technique is used for feature selection. The ANN-QCAR prediction models are then developed to link the compositions along with its weight to their reported biological activity.

GAs are general-purpose evolutionary algorithms that can be used for optimization [16]. In a GA, each population member is a potential solution, which is equal to its population size. GAs were first introduced by Holand [17] which is a search algorithm. These are stochastic optimization method and

provide a powerful technology to perform random searches in a large problem space. An introduction and overview of the application of GAs are reported by Venkatasubramanian and Sundaram [18]. The wide range of studies in QSAR has been studied using GAs.

Artificial neural networks (ANNs) have been applied to nonlinear classification and forecasting problems. In an ANN model, a neuron is an elemental processing unit that forms part of a larger network. Neural networks can be applied to form basic types of applications like association, clustering, transformation and modeling. Lots of application of ANN has been applied in structural drug design.

Presently both GA and ANN were applied to predict the bioactivity of the optimized herbal drug. Based on the proposed input data set GA and ANN can be applied to predict the overall bioactivity. The algorithm is shown in figure 1.

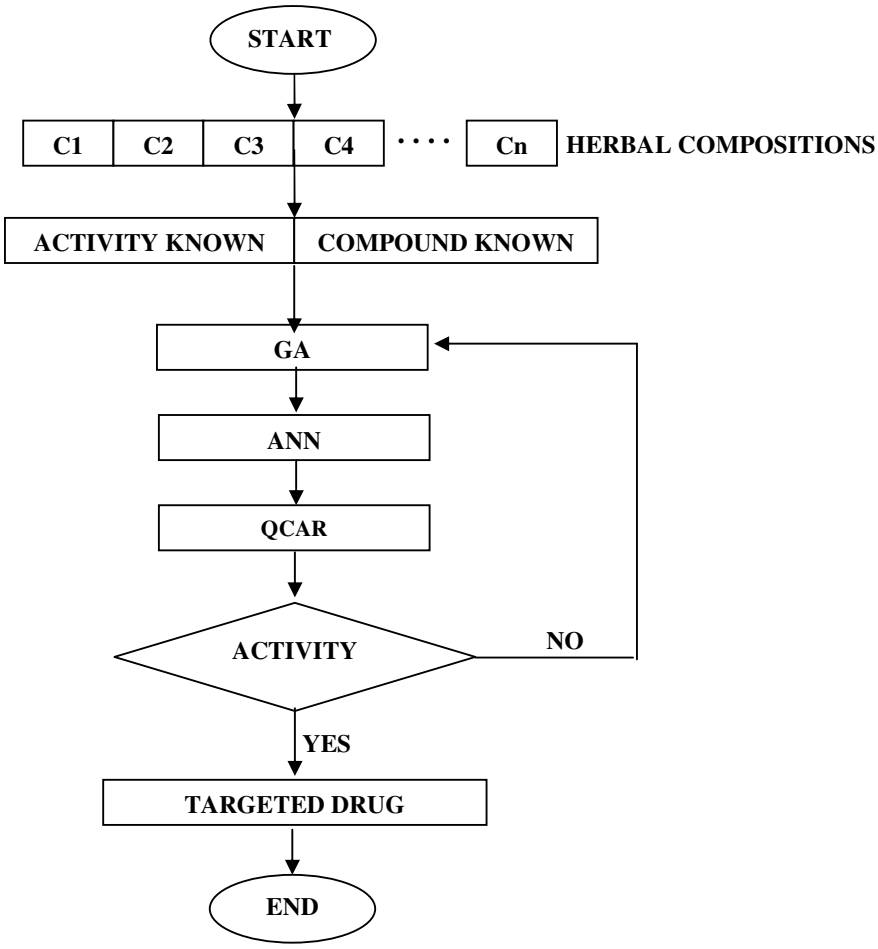


**Figure 1** Algorithm for Optimal Composition

## 4. RESULTS AND DISCUSSIONS

The algorithm was implemented using in house C programme. All calculations were carried out on a computer with a 2.0 GHz Core to Duo processor. The simulation was performed using predictive GA and integrated with ANN having one hidden layer. Table 1 shows the predicted bioactivity of the drug samples. There are 20 no of drug samples having 1 gm weight each. The weight of active compounds of each individual drug is also shown in the table. It is observed that the drug sample no 17 has the highest bioactivity which is the optimized drug. The same has been plotted in figure no 2. The compositions of the drug sample no 17 is the optimized compositions to get the highest bioactivity. The validation of this drug has yet to be done.

Herbal medicine is prevailing from ancient medical philosophy since more than thousands of years. The exact mechanism of action of herbal drug is still a controversial, widespread application of herbal product has proven the availability of herbal drug product, which demands huge workload in sample separation and bio-screening. In most of the cases, some active compositions can be discovered in this way, but the pure compound obtained cannot have the same medicinal effect as the original formulation. On the other hand, because these active components are derived from clinically useful herbal medicines, appropriate combination of active components will improve the hit rate in discovering potential herbal drug. Thus, as suggested new herbal drug should be designed as a combination of several active compositions to ensure the holistic and synergistic effects of herbal medicine. Therefore a feasible way for the production of combined herbal medicine suggests two steps. The first step is rapid and repeatable isolation of active components. The second step is combining these active components in proper ratio for its better efficacy.

In clinical practice, Indian Aurvedic practitioners often modulate the proportion of herbs according to the status of patients. In this way, the chemical composition of herbal medicine is changed. This adjustment happens according to clinical experience of the doctor. In this study, interpretation of created soft computing models can give an insight into the chemical compositions and biological action and allow for narrowing the combinatorial range of active compositions. Such focused screening can reduce the repeated experiments and increase the effectiveness of herbal drug design.

| | Major anti-diabetic active components (in gm) | | | | | | Biological activity |
|---|---|---|---|---|---|---|---|
| Sample number | C1 | C2 | C3 | C4 | C5 | C6 | Reduction in blood glucose level (%) |
| 1 | 0.285 | 0.232 | 0.000 | 0.068 | 0.175 | 0.240 | 71.38 |
| 2 | 0.105 | 0.208 | 0.111 | 0.266 | 0.056 | 0.254 | 73.37 |
| 3 | 0.241 | 0.108 | 0.000 | 0.222 | 0.105 | 0.324 | 74.47 |
| 4 | 0.308 | 0.115 | 0.117 | 0.161 | 0.182 | 0.117 | 74.87 |
| 5 | 0.079 | 0.085 | 0.299 | 0.112 | 0.291 | 0.134 | 75.53 |
| 6 | 0.198 | 0.215 | 0.000 | 0.172 | 0.268 | 0.147 | 76.85 |
| 7 | 0.148 | 0.252 | 0.210 | 0.088 | 0.249 | 0.053 | 78.38 |
| 8 | 0.000 | 0.052 | 0.289 | 0.275 | 0.185 | 0.199 | 78.54 |
| 9 | 0.168 | 0.233 | 0.000 | 0.228 | 0.115 | 0.256 | 79.01 |
| 10 | 0.138 | 0.242 | 0.085 | 0.172 | 0.291 | 0.092 | 79.47 |
| 11 | 0.239 | 0.134 | 0.221 | 0.181 | 0.000 | 0.225 | 79.78 |
| 12 | 0.128 | 0.324 | 0.000 | 0.274 | 0.192 | 0.082 | 80.26 |
| 13 | 0.253 | 0.104 | 0.288 | 0.172 | 0.082 | 0.101 | 80.40 |
| 14 | 0.170 | 0.000 | 0.150 | 0.330 | 0.180 | 0.170 | 81.29 |
| 15 | 0.165 | 0.155 | 0.175 | 0.225 | 0.135 | 0.145 | 82.36 |
| 16 | 0.000 | 0.210 | 0.200 | 0.190 | 0.250 | 0.150 | 83.43 |
| 17 | 0.330 | 0.140 | 0.180 | 0.170 | 0.180 | 0.000 | 86.71 |
| 18 | 0.900 | 0.120 | 0.150 | 0.180 | 0.210 | 0.250 | 86.43 |
| 19 | 0.135 | 0.145 | 0.155 | 0.165 | 0.175 | 0.225 | 86.67 |
| 20 | 0.145 | 0.175 | 0.180 | 0.150 | 0.160 | 0.190 | 86.66 |

**Table 1:** Predicted bioactivity of individual optimized drugs

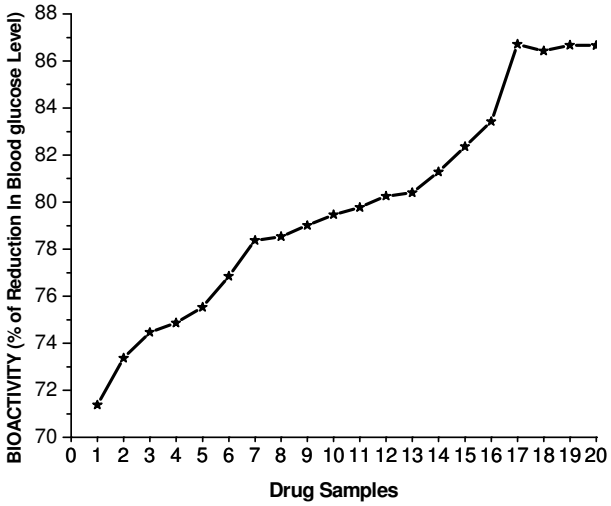


**Figure 2:** The line symbol graph showing the relation between the drug sample and their maximum bioactivity. The X-axis represents the drug sample number and the Y-axis represents the simulated bioactivities.

## 5. CONCLUSIONS

The present work is just a preliminary one to study only the concept of drug design using soft computing approach. Further work will be extended for the development of drug, which can reduce, both time and cost. The hybrid soft computing approach can be extended for any kind of diseases. These models can be useful for predicting the biological activity of new untested drug for identifying new active compounds in the traditional drug.

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# The Process of Information Extraction through Natural Language Processing

**S Acharya** sandigdhaacharya@yahoo.co.in
*Lecturer/CSE Deptt,*
*Synergy Institute of Technology*
*Bhubaneswar,Orissa,India,752101*

**S Parija** smita.parija@gmail.com
*Asst.Prof/ECE Deptt,*
*Synergy Institute of Technology.*
*Bhubaneswar, Orissa, India, 752101*

**Abstract**

Information Retrieval (IR) is the discipline that deals with retrieval of unstructured data, especially textual documents, in response to a query or topic statement, which may itself be unstructured, e.g., a sentence or even another document, or which may be structured, e.g., a Boolean expression. The need for effective methods of automated IR has grown in importance because of the tremendous explosion in the amount of unstructured data, both internal, corporate document collections, and the immense and growing number of document sources on the Internet.. The topics covered include: formulation of structured and unstructured queries and topic statements, indexing (including term weighting) of document collections, methods for computing the similarity of queries and documents, classification and routing of documents in an incoming stream to users on the basis of topic or need statements, clustering of document collections on the basis of language or topic, and statistical, probabilistic, and semantic methods of analyzing and retrieving documents. Information extraction from text has therefore been pursued actively as an attempt to present knowledge from published material in a computer readable format. An automated extraction tool would not only save time and efforts, but also pave way to discover hitherto unknown information implicitly conveyed in this paper. Work in this area has focused on extracting a wide range of information such as chromosomal location of genes, protein functional information, associating genes by functional relevance and relationships between entities of interest. While clinical records provide a semi-structured, technically rich data source for mining information, the publications, in their unstructured format pose a greater challenge, addressed by many approaches.

**Keywords:** Natural language Processing(NLP),Information retrieval, Text Zoning

## 1. INTRODUCTION

Natural Language Processing (NLP) [1]is the computerized approach to analyzing text that is based on both a set of theories and a set of technologies, and being a very active area of research and

development, there is not a single agreed-upon definition that would satisfy everyone, but there are some aspects, which would be part of any knowledgeable person's definition.

Definition: Natural Language Processing is a theoretically motivated range of computational techniques for analyzing and representing naturally occurring texts at one or more levels of linguistic analysis for the purpose of achieving human-like language processing for a range of tasks or applications. Several elements of this definition can be further detailed. Firstly the imprecise notion of 'range of computational techniques' is necessary because there are multiple methods or techniques from which to choose to accomplish a particular type of language analysis. 'Naturally occurring texts' can be of any language, mode, genre, etc. The texts can be oral or written. The only requirement is that they be in a language used by humans to communicate to one another. Also, the text being analyzed should not be specifically constructed for the purpose of the analysis, but rather that the text be gathered from actual usage.

The notion of 'levels of linguistic analysis' (to be further explained in Section 2) refers to the fact that there are multiple types of language processing known to be at work when humans produce or comprehend language. It is thought that humans normally utilize all of these levels since each level conveys different types of meaning. But various NLP systems utilize different levels, or combinations of levels of linguistic analysis, and this is seen in the differences amongst various NLP applications. This also leads to much confusion on the part of non-specialists as to what NLP really is, because a system that uses any subset of these levels of analysis can be said to be an NLP-based system. The difference between them, therefore, may actually be whether the system uses 'weak' NLP or 'strong' NLP. 'Human-like language processing' reveals that NLP is considered a discipline within Artificial Intelligence (AI). And while the full lineage of NLP does depend on a number of other disciplines, since NLP strives for human-like performance, it is appropriate to consider it an AI discipline. 'For a range of tasks or applications' points out that NLP is not usually considered a goal in and of itself, except perhaps for AI researchers. For others, NLP is the means for 1 Liddy, E. D. In Encyclopedia of Library and Information Science, 2nd Ed. Marcel Decker, Inc. accomplishing a particular task. Therefore, you have Information Retrieval (IR) systems that utilize NLP, as well as Machine Translation (MT), Question-Answering, etc. The goal of NLP as stated above is "to accomplish human-like language processing". The choice of the word 'processing' is very deliberate, and should not be replaced with 'understanding'. For although the field of NLP was originally referred to as Natural Language Understanding (NLU) in the early days of AI, it is well agreed today that while the goal of NLP is true NLU, that goal has not yet been accomplished. A full NLU System would be able to:

1. Paraphrase an input text
2. Translate the text into another language
3. Answer questions about the contents of the text
4. Draw inferences from the text

While NLP has made serious inroads into accomplishing goals 1 to 3, the fact that NLPsystems cannot, of themselves, draw inferences from text, NLU still remains the goal of NLP. There are more practical goals for NLP, many related to the particular application for which it is being utilized. For example, an NLP-based IR system has the goal of providing more precise, complete information in response to a user's real information need. The goal of the NLP system here is to represent the true meaning and intent of the user's query, which can be expressed as naturally in everyday language as if they were speaking to a reference librarian. Also, the contents of the documents that are being searched will be represented at all their levels of meaning so that a true match between need and response can be found, no matter how either are expressed in their surface form.

## 2. INFORMATION EXTRACTION

*What is Information Extraction?*

This volume takes a broad view of information extraction [2] as any method for filtering information from large volumes of text. This includes the retrieval of documents from collections and the tagging of particular terms in text. In this paper we shall use a narrower definition: the identification of instances of a

particular class of events or relationships in a natural language text, and the extraction of the relevant arguments of the event or relationship. Information extraction there- fore involves the creation of a structured representation (such as a data base) of selected information drawn from the text.

The idea of reducing the information in a document to a tabular structure is not new. Its feasibility for sublanguage texts was suggested by Zellig Harris in the 1950's, and an early implementation for medical texts was done at New York University by Naomi Sager [5]. However, the specific notion of information extraction described here has received wide currency over the last decade through the series of Message Understanding Conferences [1, 2, 3, 4, 14].We shall discuss these Conferences in more detail a bit later, and shall use simplified versions of extraction tasks from these Conferences as examples throughout this paper the type of attack (bombing, arson, etc.), the date, location, perpetrator (if stated), targets, and effects on targets. Other examples of extraction tasks are international joint ventures (where the arguments included the partners, the new venture, its product or service, etc.) and executive succession (indicating who was hired or _red by which company for which position).

Information extraction is a more limited task than \full text understanding". In full text understanding, we aspire to represent in a explicit fashion all the information in a text. In contrast, in information extraction we delimit in advance, as part of the specification of the task, the semantic range of the output: the relations we will represent, and the allowable _leers in each slot of a relation. Identify specific pieces of information (data) in a unstructured or semi-structured textual document. Transform unstructured information in a corpus of documents or web pages into a structured database.

Applied to different types of text:

–Newspaper articles
–Web pages
–Scientific articles
–Newsgroup messages
–Classified ads
–Medical notes

In many application areas of text analysis, for instance, in information retrieval and in text mining, shallow representations of texts have been recently widely used. In in- formation retrieval, such shallow representations allow for a fast analysis of the in- formation and a quick respond to the queries. In text mining, such representations are used because they are easily extracted from texts and easily analyzed. Recently in all text-oriented applications, there is a tendency to begin using more complete representations of texts than just keywords, i.e., the representations with more types of textual elements. For instance, in information retrieval, these new representations increase the precision of the results; in text mining, they ext end the kinds of discovered knowledge.Many web pages are generated automatically from an underlying database. Therefore, the HTML structure of pages is fairly specific and regular However, output is intended for human consumption, not machine interpretation. An IE system for such generated pages allows the web site to be viewed as a structured database.

### 2.1 Techniques of Information Retrival

#### 2.1.1 Text Zoning

Turns a text into a set of useful text segment(like headers,paragraphs,table) May be topic based using keywords or static. Depends on the structure of the text in domain of application. Discard unwanted segment of text.

#### 2.1.2 Preprocessing

Take as input a stream of characters carried out tokenisation & sentence segmentations(convert txt segment into a sequence of sentence,disambiguaten fullstop) Part-of-speech tagging.named entity, spelling correction has been carried out.

**2.1.3 Filtering**
Throws away sentences considered to be irrelevant.Primary consideration is processing time[31,32]. Relevance decision can use manually or statisticallyderived keywords.

**2.1.4 Preparsing**
Ingoing from a sequence of words to a parse tree, some structure can be identify more reliably than other( noun,prepositional phrases,appositives) Uses finite state grammar & special word list.

**2.1.5 Name recognition entity**
Name may contain unknown words Identify of names simplify parsing. IE templates slots are typically filled with name.

Temporalexpreexpression(time,date,duration).Numberexpression(number,money,measure,sped,volume,temprature,percentage,cardinal).Simpleregularexpression(postalcode,studentid,telephoneno.)Entityname (person,organization,location)

**2.1.6 Parsing**
Takes as inputa sequence of lexical items and smallscale structure built by the parser.Produce as output a set of parse tree fragments, correspond to subsentential unit.Goal is to detrmine the major elements in the sentence (nounphrase,verbphrase)

**2.1.7 Fragment combination**
Take as input a set of parse tree fragments derived from a sentences.Tries to combine fragments into a representation for he entire sentence[6].

**2.1.9 Semantic interpretation**
Generate a semantic structure or logical form or event frame from a parse tree or a collection of parse tree fragment. What is a semantic structure An explicit representation of the relationship between participant in sentence. Goal is to map syntatic structure into structure that encode information relevance template filling[7].

**2.1.10 Lexical disambiguation**
Turns a semantic structure with ambigous predicate into unambigous predicate.This task may be carried out in a number of places in a system.In restricted domains this may not be an issue –the one sense per document assumption. Only one sense of the word is used in the complete domain.

**2.1.11 Coreference Resolution**

- Identify different description of he same entity in different parts of text and relates them in some way. identify,meronymy.,reference to events.

- Techniques number & gender agreement for pronoun(Ram met Shyam,he later stated.) semantic consistency based on taxonomic information(toyota motor corp.”the japenese automoter”.) some notion of focus(pronoun typically refer to something mentioned in the previous sentence).

**2.1.12Template generation**
Derive final output templates from the semantic structures.Carries out lowlevel formatting and normalization of data.

**2.1.13 Evaluations**

Precision=Ncorrect/Nresponse,

Recall=Ncorrect/Nkey

F= (2*precision*recall)/(precision+recall)

## 3. OBSERVATIONS

### 3.1. Strengths of SVM

- the solution is unique
- the boundary can be determined only by its support vectors, namely SVM[33] is robust against changes of all vectors but its support vectors
- SVM is insensitive to small changes of the parameters different SV classifiers constructed by using different kernels (polynomial, RBF, neural net) extract the same support Vectors**.**

**Weaknesses of SVM**

- It takes more time.
- SVM is used only for categorization of documents and user feedback

### 3.2 Conceptual Graph:

- It is more accurate than other two model.
- structured semantic matching can improve both recall and precision
- . Each relation associated with the entry induces a subgraph
- There are many graph to derive.

### 3.3 Boolean Retrieval:

- Process large document quickly.
- Allow more flexible matching.
- Need invarted index files.
- Used more memory space.
- More time take to access

| | Standard Boolean |
|---|---|
| Goal | • capture Conceptual structure and Contextual Information |
| Methods | • Coordination:AND,OR,NOT<br>• Proximity<br>• Fields<br>• Stemming/Truncation |
| (+) | • Easy to implement<br>• Computationally Efficient<br>=all the major online databases use it.<br>• Expressiveness and Clarity<br>Synonm specifications (OR –Clauses) and phrases (AND –Clauses) |
| (-) | • Difficult to Construct Boolean queries<br>• All or Nothing.<br>ANDBtoo severe ,and OR does not differentiate<br>Enough<br>• .Difficult to control output:Null output↔Overload.<br>• No Ranking.<br>• No weighting of index or query terms.<br>• No uncertainty measure. |

**FIGURE 1.** Boolean Retrieval

| Parameters | Boolean Retrieval | Conceptual Graph | SVM |
|---|---|---|---|
| Types of Solution | More solutions | More solutions | Unique Solutions |
| Data Types | Linear Documents | Linear Documents | Multidimensional data |
| Performance | More accurate | Most accurate than other two models | More accurate |
| Evaluation | Simple | Simple | More complex |
| Space Complexity | Uses more memory | Uses more memory | Uses less memory |
| Efficiency | Retrieves large documents quickly | Retrieval of information | Categorization of documents |

**FIGURE 2.** Comparision Study.

## 4. SURVEY ON PAPERS:

**4.1** Paper**1**

The combined use of linguistic ontologies and structured semantic matching is one of the promising ways to improve both recall and precision. In this paper, we propose an approach for semantic search by matching conceptual graphs. The detailed definitions of semantic similarities between concepts, relations and conceptual graphs are given. According to these definitions of semantic similarity, we propose our conceptual graph matching algorithm that calculates the semantic similarity. The computation complexity of this algorithm is constrained to be polynomial. A prototype of our approach is currently under development with IBM China Research Lab

**4.1 Paper2**

As defined in this way, information retrieval used to be an activity that only a few people engaged in: reference librarians, paralegals, and similar professional searchers. Now the world has changed, and hundreds of millions of people engage in information retrieval every day when they use a web search engine or search their email.1 Information retrieval is fast becoming the dominant form of information access, overtaking traditional database style searching (the sort that is going on when a clerk says to you: “I’m sorry, I can only look up your order if you can give me your order ID”). Information retrieval can also cover other kinds of data and information problems beyond that specified in the core definition above. The term “unstructured data” refers to data that does not have clear, semantically overt, easy-for-a-computer structure. It is the opposite of structured data, the canonical example of which is a relational database, of the sort companies usually use to maintain product inventories and personnel records. In reality, almost no data are truly “unstructured.” This is definitely true of all text data if you count the latent linguistic structure of human languages. But even accepting that the intended notion of structure is overt structure, most text has structure, such as headings, paragraphs, and footnotes, which is commonly represented in documents by explicit markup (such as the coding underlying web pages). Information retrieval is also used to facilitate “semi structured”. search such as finding a document where the title contains Java and the body contains threading. The field of IR also covers supporting users in browsing or filtering document

collections or further processing a set of retrieved documents. Given a set of documents, clustering is the task of coming up with a good grouping of the documents based on their contents. It is similar to arranging books on a bookshelf according to their topic. Given a set of topics, standing information needs, or other categories (such as suitability of texts for different age groups), classification is the task of deciding which class(es), if any, each of a set of documents belongs to. It is often approached by first manually classifying some documents and then hoping to be able to classify new documents automatically. Information retrieval systems can also be distinguished by the scale at which they operate, and it is useful to distinguish three prominent scales. In *web search*, the system has to provide search over billions of documents stored on millions of computers. Distinctive issues are needing to gather documents for indexing, being able to build systems that work efficiently at this enormous scale, and handling particular aspects of the web, such as the exploitation of hypertext and not being fooled by site providers manipulating page content in an attempt to boost their search engine rankings, given the commercial importance of the web. We focus on all these issues in

Chapters 19–21. At the other extreme is *personal information retrieval*. In the last few years, consumer operating systems have integrated information retrieval (such as Apple's Mac OS X Spotlight or Windows Vista's Instant Search). Email programs usually not only provide search but also text classification: they at least provide a spam (junk mail) filter, and commonly also provide either manual or automatic means for classifying mail so that it can be placed directly into particular folders. Distinctive issues here include handling the broad range of document types on a typical personal computer, and making the search system maintenance free and sufficiently lightweight in terms of startup, processing, and disk space usage that it can run on one machine without annoying its owner. In between is the space of *enterprise, institutional,* and *domain-specific search*, where retrieval might be provided for collections such as a corporation's internal documents, a database of patents, or research articles on biochemistry. In this case, the documents are typically stored on centralized file systems and one or a handful of dedicated machines provide search over the collection. This book contains techniques of value over this whole spectrum, but our coverage of some aspects of parallel and distributed search in web-scale search systems is comparatively light owing to the relatively small published literature on the details of such systems. However, outside of a handful of web search companies, a software developer is most likely to encounter the personal search and enterprise scenarios. In this chapter, we begin with a very simple example of an IR problem,Central inverted index data structure.We then examine the Boolean retrieval model and how Boolean queries are processed .

**An example information retrieval problem**

A fat book that many people own is *Shakespeare's Collected Works*. Suppose you wanted to determine which plays of Shakespeare contain the words Brutus and Caesar and not Calpurnia. One way to do that is to start at the beginning and to read through all the text, noting for each play whether it contains Brutus and Caesar and excluding it from consideration if it contains Calpurnia. The simplest form of document retrieval is for a computer to do this sort of linear scan through documents. This process is commonly grep referred to as grepping through text, after the Unix command grep, which performs this process. Grepping through text can be a very effective process, especially given the speed of modern computers, and often allows useful possibilities for wildcard pattern matching through the use of regular expressions. With modern computers, for simple querying of modest collections (the size of *Shakespeare's* Collected Works is a bit under one million words of text in total), you really need nothing more. But for many purposes, you do need more 1. To process large document collections quickly. The amount of online data has grown at least as quickly as the speed of computers, and we would now like to be able to search collections that total in he order of billions to trillions of words. 2. To allow more flexible matching operations.

## 5. CONCLUSION

After the survey of these two[33,34] papers we observed that the above models provides the best match. But we need the exact match for Information Extraction. Hence there is a further need of another model which should provide the exact match .The paper presented two approaches to extracting information Structures. While the manual approach is more accurate and can be engineered in a domain specific

way, the automated approach is also advantageous because of its scalability.Our evaluation of the automated system on both the BioRAT[25] and GeneWays[23] datasets shows that our system performs comparably with other existing systems. Both the systems compared were built by manual rule engineering approach, and involve a repetitive process of improving the rules which take up a lot of effort and time. Our system is able to achieve similar results with minimal effort on part of the developer and user. While advantageous on this aspect, we realize that our system is also in need of improvements in tagging entities to boost the performance. Improvements in the interaction extractor module will also bring up the precision of the system. Nevertheless, we have proven that a syntactic analysis of the sentence structure from full sentence parses produces results comparable to many of the existing systems for interaction extraction[[29,30].

Semantic matching has been raised for years to improve both recall and precision of information retrieval.[9,14] finds first the most similar entities and then observe the correspondence of involved relationship. However, with this kind of simplification, matching on nodes is separate without the organization of sub graphs. In contrary, we try to retain sub graph structure in our matching procedure with as less cost as possible. OntoSeek [8,13] defines the match on isomorphism between query graph and a sub graph of resource graph where the labels of resource graph should be subsumed by the corresponding ones of query graph. The strict definition of match makes their system can't support partial matching. The assumption that user would encode the resource descriptions by hand also limits its popularization. Different from it, our method not only supports the partial matching but also introduces the weight to reflect user's preferences, which makes our method more flexible. Some previous work have discussed the issue of semantic distance, such as [5] [14] and [15]. The basic thought is to define the distance between two concepts as the number of arcs in the shortest path between two concepts in the concept hierarchy which does not pass through the absurd type. [14] modified the definition and defined the distance as the sum of the distances from each concept to the least concept which subsumes the two given concepts. We adopt their original thought and make some modifications to make it suitable to our work.

The measurement of concept similarity was also studied before. [17] builds their similarity definition on the information interests shared by different concepts, while [16] defines the similarity between two concepts as the information content (entropy) of their closest common parent, and besides take the density in different parts of the ontology into account. The measuring of concept similarity in our approach is different from them and is simpler. Of course, our approach is far from perfect. It needs further study based on collected experiment data in the future .Now a days, with the electronic information explosion caused by Internet, increasingly diverse information is available. To handle and use such great amount of information, improved search engines are necessary. The more information about documents is preserved in their formal representation used for information retrieval, the better the documents can be evaluated and eventually retrieved. Based on these ideas, we are developing a new information retrieval system. This system performs the document selection taking into account two different levels of document representation. The first level is the traditional keyword document representation. It serves to select all documents potentially related to the topic(s) mentioned in the user's query. The second level is formed with the conceptual graphs[20,21,22] reflecting some document details, for instance, the document intention. This second level complements the topical information about the documents and provides a new way to evaluate the relevance of the document for the query. the query and extracts from it a list of topics (keywords). The keyword search finds all relevant documents for such a keyword-only query. Then, the information extraction module constructs the conceptual graphs of the query and the retrieved documents, according to the process described in section 3. This information is currently extracted from titles [10] and abstracts [11] of the documents. These conceptual graphs describe mainly the intention of the document, but they can express other type of relations, such as cause-effect relations [12,13].

This graph indicates that the document in question has the intention of demonstrating the validity of the technique Then the query conceptual graph is compared – using the method described in this paper – with the graphs for the potentially relevant documents. The documents are then ordered by their value *s* of the similarity to the query. After this process the documents retrieved at the beginning of the list will not only mention the key-topics expressed in the query, but also describe the intentions specified by the user. This technique allows improving the retrieval of information in two main directions:

1.It permits to search the information using not only topical information, but also extratopical, for instance, the document intentions.

2. It produces a better raking of those documents closer to the user needs, not only in terms of subject. We have described the structure of an information retrieval system that uses the comparison of the document and the query represented with conceptual graphs to improve the precision of the retrieval process by better ranking on the results. In particular, we have described a method for measuring the similarity between conceptual graph representations of two texts. This method incorporates some well-known characteristics, for instance, the idea of the Dice coefficient – a widely used measure of similarity for the keyword representations of texts. It also incorporates some new characteristics derived from the conceptual graph structure, for instance, the combination of two complementary sources of similarity: the conceptual similarity and the relational similarity. This measure is appropriate for text comparison because it considers not only the topical aspects of the phrases (difficult to obtain from short texts) but also the relationships between the elements mentioned in the texts. This approach is especially good for short texts. Since in information retrieval, in any comparison operation at least one of the two elements, namely, the query, is short, our method is relevant for information retrieval. Currently, we are adapting this measure to use a concept hierarchy given by the user, i.e. an *is-a* hierarchy, and to consider some language phenomena as, for example, synonymy. However, the use of the method of comparison of the texts using their conceptual graph representations is not limited by information retrieval. Other uses of the method include text mining and document classification.

The information extraction system based on complex sentence processing is able to handle binary relations between genes and proteins, and some nested relations. However, researchers are also interested in contextual information such as the location and agents for the interaction and the signaling pathways of which these interactions are a part. Our tasks for future work include the following

- Handling negations in the sentences (such as ”not interact”, ”fails to induce”, ”does not inhibit”)
- Identification of relationships among interactions extracted from a collection of simple sentences (such as one interaction stimulating or inhibiting another)
- Extraction of detailed contextual attributes (such as bio-chemical context or location) of interactions and
- Building a corpus of biomedical abstracts and extracted interactions that might serve as a benchmark for related extraction systems. Attempts to improve the parse output of the Link Grammar System were also undertaken. The dictionaries of the Link Grammar Parser[18] were augmented with medical terms with their linking to polynomial.

Before discussing the complexity of the algorithm, we firstly consider the effect caused by cycles in requirements provided by Szolovit in his website. In spite of the improvement in performance of the Link Grammar Parser, this approach was discontinued in favor of the Pre-processor subsystem because of the increase in time taken to load the dictionaries and for parsing. Semantic[24] analysis based on proposed information extraction techniques would enable automated extraction of detailed gene-gene relationships, their contextual attributes and potentially an entire history of possibly contradictory sub-pathway theories from biomedical abstracts in PubMed thus allowing our system to generate more relevant and detailed recommendations. When applying graph matching algorithm, the greatest worry comes about the computation complexity, since it is well known that Maximum Sub graph Matching is a NP-complete problem. Fortunately, it can be expected in our algorithm that the computation complexity will be constrained graphs to our algorithm. Since the algorithm is recursive, the cycle in graph will lead to an unending recursion and will be fatal to our algorithm. So we must eliminate the cycles in graphs before we match them. We can handle it simply by duplicating the concept in cycles. Surely, this will increase the computation complexity, especially when the cycle is very complex. Fortunately, benefiting from the domain specific characters, cycles in graphs are very rare especially in commodity domain. So we ignore it here.In the following, we will discuss the complexity of our algorithm. Since cycles in graphs are very rare and the cycles can be eliminated simply, we will only concern the tree structure. Without losing generality, we can suppose that the query graph and the resource graph contain n arcs each and are both *l*-branch trees of *i* height, so there are more than *li* relations. We use C(*i*) to denote the time

complexity of matching two trees both of *i* height. As shown in the algorithm, we will calculate the similarity between the two entries firstly We use a constant c to represent the time spent in calculating concept similarity. After this step, the time complexity is c; then we need to calculate the similarity between each sub graph pair. Since each entry will induce *l* sub graphs we need *l*2 times recursive invocations. These sub graphs are all *l*-branch trees of *i*-1 height, so in every invocation, the time complexity is C(*i*-1). Here we ignore the time to calculate similarity between relations. After these two loops, the time complexity will be c+*l*2*C(*i*-1). Once we determine the similarity between each sub graph pair, we should find out the best match from different mate combinations There exists *l*! combinations in these *l*2 sub graph pairs, so how to handle it efficiently is important. We translate the issue into a *maximum flow* problem and execute Bellman-Ford[26] algorithm *l* times to solve it4, whose computation complexity is *l*3, and the cumulative complexity is *l*4. So the complexity can be described as follows: From the formula, we can see that C(*i*) is about *l2i+2*. Generally, when *l* is not very small, the number of arcs n will approximate *li*, so the complexity will be n*2l2*. If *l*<<n, the complexity will be O(n2). For the worst case, suppose there is only 1 layer in the query graph, i.e. *l*=n, the complexity is O(n4). Since the algorithm combines syntactic and semantic context information in the whole process[27,28],the advantages over traditional keyword match technique can easily be seen. For example, a description is about 'soft collar shirt' and another is about 'soft shirt with straight collar'. They are both selected by keyword search when using 'soft 4 InThe initial observation in this paper is that binary decisions are not good enough for ontology evaluation, when hierarchies are involved. We propose an Augmented Precision and Recall measure that takes into account the ontological distance of the response to the position of the key concepts in the hierarchy**.**

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