7.1 Flow of Algorithm

The main motivation behind using the wavelet analysis for speech processing is, wavelets have good localization in time and frequency domain. Wavelets are the functions suited to the expansion of non-stationary continuous signals. Wavelet transform maps signal from time domain into time-frequency domain. Redundancy and irrelevancy present in the speech signal (quasi-stationary signal) can be easily removed using wavelet transform.

Figure 7.1 Block diagram of algorithm

Above Figure 7.1 shows the flow of the speech enhancement algorithm. For achieve desired modification six step procedure is required on the input speech signal.
**Step 1: Take Reference Speech from Adaptive Algorithm to be Processed**

As discussed in above chapters as per the requirement any adaptive algorithm can be selected for the noise reduction. Again after reduction of noise filtered speech should be trained according to the format of the audiogram. Person to person modification of band will change. So as per the requirement for the patient process should be carried out. In that band enhancement is necessary. Filtered speech should be passed to wavelet algorithm for further process.

1. Set the property of the speech signal
2. Get output of adaptive algorithm for further process in the enhancement.
3. Study audiogram of specific person for modification decision

**Step 2 Implementation of Discrete Wavelet Transform – MRA**

![Forward discrete wavelet transform](image)

**Figure 7.2 Forward discrete wavelet transform**

A discrete wavelet transform performs a multistage signal decomposition using filter bank structure. DWT coefficients can be calculated using Mallat’s fast tree algorithm also...
known as fast wavelet transform (FWT). This algorithm consists of number of identical stages. At each stage input signal is filtered through a low pass and high pass filter. The filtered samples are then down sampled by a factor of two. The output of low pass filter gives coarser information and output of high pass filter gives detail information. In normal DWT, coarse data from previous stage is used as input of next stage. This gives progressively low frequency resolution and details.

A window of input samples \( f(n) \) is applied at the input of filter bank. The size of window is power of two (If it is not power of two, some algorithm is applied to divide this samples into different blocks and ensure that each block length is power of two). First some of the samples are taken in the form small window for the processing. That small number of samples set is processed in parallel. That data is one time get convoluted with high pass filter coefficient and second time with low pass filter coefficient. As a result obtained samples are double in number. So maintain hierarchy of process it is necessary to reduce number of samples by the process of down conversion. This process of filtering and down sampling is collectively known as decimation. The down sampled output of the first high pass filter (decimated output) becomes “level-1” wavelet coefficients, which contains detail part of the signal. The output of the low pass filter is approximate output at “level-1” that can be further down sampled and given to next identical stage as shown in the Figure 7.2. From this signal, further more approximated signal and wavelet coefficient at “level-2” can separated out. By process repetition up to \( N^{th} \) level. Length of input signal is divides by two every time as level increases. Higher-level wavelet coefficients have lower characteristics time resolution and narrower characteristics bandwidth information.

In case of a forward wavelet transform, down sampling is important to ensure the size of the output samples same as input samples. Without reducing number of samples original data of them matrix would be increases and it is known as data explosion. As a byproduct of wavelet coefficient and data convolution two set of matrix can be achieved name high pass filter and low pass filter set. High pass filter matrix represents wavelet coefficients and normal value which can be known as a average can be represents by set of low pass
filter set. Increment and decrement in the samples represents time resolution and input signal and with time how signal can be changes. Gradually per discreet level number of coefficient would be decreases. Generally if number of sampled value in the matrix is 2028 as a input values first level carries 1024 wavelet values , 512 values taken by second level and 256 individual number taken by third level and so on hierarchy progresses. Each level value matrix represents unique feature of the signal. As analysis says, level by level it is two fold decrease in no of samples. Wavelet values on each level describe different number and different value so time resolution can be obtained. Moreover more time resolution wants more number of values in the level. In fact each level gives time frequency information of that small window.

1. Using command get selection of wavelet name followed by selection and detail and approximate coefficient values.
2. Detail and approximate values are averaged at some threshold.
3. Use matrix operations to get rows and columns of signal in required format.
4. Initialize the vector to store number of coefficient after each and every decomposition.
5. Get the convolution of high pass and low pass filter with original signal individually and store data of each convolution after down sampling by s using function dyadown.
6. Repeat the above step eight times.
7. At each successive executions consider convolution of high pass filter and signal as a main signal for next execution.
8. Store the value of wavelet coefficient for eight decomposition stages and no of coefficient after each stage in two different variables WC and LC.

**Step 3 Speech Enhancement**

Speech enhancement is required as per the conclusion form the audiogram of person. Band selection from the audiogram is necessary for increment in the loudness level in decibels. After selecting particular band same is selected for the modification. Normal hearing hearing threshold level is 20db in general condition for human being. Whatever
decrement is there in loudness level that required to be corrected for that frequency band it can be observed from the audiogram.

1. Select particular band of stored wavelet coefficient for the enhancement.
2. Multiply with suitable numerical value as per the required gain and selected wavelet coefficient band as per the frequency of the input speech.

**Step 4 Implementation of Inverse Discrete Wavelet Transform**

The inverse wavelet transform works exactly reverse then forward wavelet transform. Final average and $N^{th}$ level wavelet coefficients are up-sampled. Logical process handles by padding zeros between samples. As a result new matrix will be formed which is supposed pass through inverse low pass filter and inverse high pass filter which is also known as interpolation. Resultant outputs are then summed up and forwarded to next step of process. Describe process continue until end.

Final average

![Inverse discrete wavelet transform](image)

**Figure 7.3 Inverse discrete wavelet transform**

By the understanding of wavelet transform implementation for forward and reverse is not giving much pain. But in the practice real data processing arises various amount of problems in the implementation. Real image and speech carries huge amount of sampled values in the form of matrix or in series of matrix. At the time of processing it is not appropriate to apply filtering coefficient on the whole data input. Naturally windowing is must on the large amount of data. Most of the time dealing with boundaries in the
windowing generates many crucial issues. As boundaries of the window are not handled carefully, reconstruction of the data is troublesome. Generally ringing is generated at the end of the window. To solve the issue of end conditions many methods can be apply to smoothen the windowed data. Symmetric extention, zero padding and circular convolution are most popular methods. All the digital filters must have designing criteria with condition that it must have perfect reconstruction ability. Its ony true when at the time of the windowing are where samples are separated must be deal perfectly in form of window edges. Edges should be taken care such a way that any discrepancy like distortion or aliasing should not take place. Reconstructed signal must be identical to original signal. Choosing of mother curve for wavelet processing take many trade off. Directly in the time domain mother wavelet can be convolved with the original input signal. In the original process first mother wavelet is selected and from that same similar set of high pass and low pass filter can be chosen.

1. Using function ‘wfilters’ get high pass and low pass coefficient by giving specific wave name and time.
2. Normalize high pass and low pass the coefficient of wavelet.
3. Use matrix operations to get rows and columns of signal in required format.
4. Initialize the vector to store number of modified wavelet coefficient.
5. Get the convolution of high pass and low pass filter with up sampled approximate and detailed coefficients of modified wavelet individually and add them
7. Store addition and in variable ‘rS’ and repeat the procedure up to last stage reach.

**Step 5: Comparison of Modified and Original Speech Signal.**
For observing comparison between original and modified speech files both files should be reconstructed in the same wave format. Reconstructed data should be plot in graph showing frequency vs decibel. By reading the graphs enhancement in selected band can be seen as per the requirement. Power spectral density function is used to plot the frequency vs decibel graph of the two speech files named ‘pwelch’. Procedure for executing power spectral density by ‘pwelch’ algorithm
1. Fix the size of FFT window for samples.  
2. Initialize N for N point FFT.  
3. Calculate number of overlapping samples in the window.  
4. Read the original speech file and store the bytes in variable.  
5. Apply ‘pwelch’ function on the sample bytes and it returns average spectrum of signal.  
7. Convert average spectrum value in the decibel and plot the decibels vs frequency.  
7. Repeat steps five and six for the modified speech file reconstructed after IDWT.  
8. Take the difference in the decibels between two average spectrum which are in decibel.  
9. Plot the difference in decibel which gives modification for selected band.

![Spectral Analysis](image)

**Figure 7.4  Graph of power spectral density vs frequency**

Above figure 7.4 utilizes below mentioned parameters and protocols for finding out power spectral density.  
1. By default, vector is divided into eight sections with 50% overlap.  
2. Each section is windowed with a Hamming window n point FFT is applied to the windowed data and eight modified periodograms are computed and averaged.
7.2 Wavelet MRA Implementation and Separate Frequency Enhancement of Speech

Wavelet is very important tool for the enhancement of speech signal. As mentioned above for speech signal which contains information about time, frequency and magnitude, wavelet is very good approximation. Basically speech signal frequency is from in the range 300 Hz to 4 kHz. Again most important application of wavelet is in the mallat algorithm. Each and every stage of the wavelet application gives partition in the frequency domain. In simulation process stages of the wavelet application must be defined. Accordingly speech frequency band wise coefficient can be achieve. As shown in the Figure 7.5 as many number of wavelet stages are applied, more minute resolution in the frequency can be achieved. As per the requirement of the audiogram individual band of the frequency is needed to modify. For that in the wavelet coefficient band some numerical factor can be searched because of that specified decibel increment can be searched out. Because of that value some average modification is given to that band and speech can be enhanced. In the proposed simulation number of wavelet stages are eight. As a result nine band can be prepared. Band wise individual factor is selected over here because of that overall 3 dB increment results are shown in below simulation results. If more that 3 dB are required than other value of multiplying factor should be chosen.

![Figure 7.5 Sample decomposition hierarchy](image-url)
7.2.1 First Band Enhancement

Figure 7.6 Original and reconstructed speech for first band

Figure 7.7 Original and modified wavelet coefficient for first band
Wavelet multi resolution tool is very useful for frequency band partition. Figure 7.6 shows simulation results of original speech and enhanced speech. In the developed research 8 level wavelet transform is used so different nine bands are generated. First band wavelet coefficients are selected and enhancement is performed between wavelet transform and inverse wavelet transform. First band wavelet coefficients are multiplied with factor 7.67. Figure 7.7 shows simulation plot for two category of wavelet coefficient one original coefficients and other enhanced band coefficients of wavelet transform. When signal reconstructs after enhancement some magnitude variation is noticed. As per requirement any level of magnitude increment can be achieved. In present work for example 3db variation is taken. By multiplying with suitable numerical value in between with wavelet coefficient, average increment in the band can be noticed. Figure 7.8 shows psd of original band and then modified psd is overlapped on the same band also. In the simulation result second graph shows clear plot of difference in the first band for
reference. Thus, very accurate amount of sound intensity increment can be observed by taking appropriate multiplication in the MRA wavelet domain.

### 7.2.2 Second Band Enhancement

![Original and reconstructed speech for second band](image1)

![Modified speech signal](image2)

**Figure 7.9** Original and reconstructed speech for second band

![Wavelet coefficients](image3)

![Modified Wavelet coefficients](image4)

**Figure 7.10** Original and modified wavelet coefficient for second band
Using wavelet multi resolution band partition can be implemented. Figure 7.9 shows simulation results of original speech and enhanced speech. In the developed research 8 level wavelet transform is used so different nine bands are generated using wavelet db4. Second band wavelet coefficients are selected and enhancement is performed between wavelet transform and inverse wavelet transform. Second band wavelet coefficients are multiplied with factor 6.3. Figure 7.10 shows simulation plot for two category of wavelet coefficient one original coefficients and other enhanced band coefficients of wavelet transform. When signal reconstructs after enhancement some magnitude variation is noticed. As per requirement any level of magnitude increment can be achieved. In present work for example 3db variation is taken. By multiplying with suitable numerical value in between with wavelet coefficient, average increment in the band can be noticed. Figure 7.11 shows psd of original band and then modified psd is overlapped on the same band also. In the simulation result second graph shows clear plot of difference in the second band for reference. Thus, very accurate amount of sound intensity increment can be observed my taking appropriate multiplication in the MRA wavelet domain.
7.2.3 Third Band Enhancement

Figure 7.12 Original and reconstructed speech for third band

Figure 7.13 Original and modified wavelet coefficient for third band
In the shown simulation, an 8-level wavelet multi-resolution tool is used. Figure 7.12 shows simulation results of original speech and enhanced speech. Third band wavelet coefficients are selected, and enhancement is performed between the wavelet transform and inverse wavelet transform. Third band wavelet coefficients are multiplied with a factor of 1.5.

Figure 7.13 shows a simulation plot for two categories of wavelet coefficients—one representing original coefficients and the other representing enhanced band coefficients of the wavelet transform. When the signal reconstructs after enhancement, some magnitude variation is noticed. As per the requirement, any level of magnitude increment can be achieved. In the present work, for example, a 3dB variation is taken. By multiplying with a suitable numerical value between the wavelet coefficient, an average increment in the band can be noticed. Figure 7.14 shows the Power Spectral Density (PSD) of the original band and then the modified PSD is overlaid on the same band. In the simulation result, the second graph shows a clear plot of the difference in the third band for reference. Thus, very accurate sound intensity increment can be observed by taking appropriate multiplication in the MRA wavelet domain.
7.2.4 Fourth Band Enhancement

![Figure 7.15 Original and reconstructed speech for fourth band](image1)

![Figure 7.16 Original and modified wavelet coefficient for fourth band](image2)
Wavelet multi resolution tool is very useful for frequency band partition. Figure 7.15 shows simulation results of original speech and enhanced speech. In the developed research 8 level wavelet transform is used so different nine bands are generated. Fourth band wavelet coefficients are selected and enhancement is performed between wavelet transform and inverse wavelet transform. Fourth band wavelet coefficients are multiplied with factor 4.1. Figure 7.16 shows simulation plot for two category of wavelet coefficient one original coefficients and other enhanced band coefficients of wavelet transform. When signal reconstructs after enhancement some magnitude variation is noticed. As per requirement any level of magnitude increment can be achieved. In present work for example 3db variation is taken. By multiplying with suitable numerical value in between with wavelet coefficient , average increment in the band can be noticed. Figure 7.17 shows psd of original band and then modified psd is overlapped on the same band also. In the simulation result second graph shows clear plot of difference in the fourth band for reference. Thus, very accurate amount of sound intensity increment can be observed my taking appropriate multiplication in the MRA wavelet domain.
7.2.5 Fifth Band Enhancement

Figure 7.18 Original and reconstructed speech for fifth band

Figure 7.19 Original and modified wavelet coefficient for fifth band
Speech is complex wave of more than one frequency. Wavelet tool gives frequency band partition. Figure 7.18 shows simulation results of original speech and enhanced speech. In the developed research 8 level wavelet transform is used. Fifth band wavelet coefficients are selected and enhancement is performed between wavelet transform and inverse wavelet transform. Fifth band wavelet coefficients are multiplied with factor 2.6. Figure 7.19 shows simulation plot for two category of wavelet coefficient one original coefficients and other enhanced band coefficients of wavelet transform. When signal reconstructs after enhancement some magnitude variation is noticed. As per requirement any level of magnitude increment can be achieved. In present work for example 3db variation is taken. By multiplying with suitable numerical value in between with wavelet coefficient , average increment in the band can be noticed. Figure 7.20 shows psd of original band and then modified psd is overlapped on the same band also. In the simulation result second graph shows clear plot of difference in the fifth band for reference. Thus, very accurate amount of sound intensity increment can be observed my taking appropriate multiplication in the MRA wavelet domain.
7.2.6 Sixth Band Enhancement

Figure 7.21 Original and reconstructed speech for sixth band

Figure 7.22 Original and modified wavelet coefficient for sixth band
For quasi periodic speech signal wavelet multi resolution tool is very useful for frequency band partition. Figure 7.21 shows simulation results of original speech and enhanced speech. In the developed research 8 level wavelet transform is used. Sixth band wavelet coefficients are selected and enhancement is performed between wavelet transform and inverse wavelet transform. Sixth band wavelet coefficients are multiplied with factor 1.8. Figure 7.22 shows simulation plot for two category of wavelet coefficient one original coefficients and other enhanced band coefficients of wavelet transform. When signal reconstructs after enhancement some magnitude variation is noticed. As per requirement any level of magnitude increment can be achieved. In present work for example 3 db variation is taken. By multiplying with suitable numerical value in between with wavelet coefficient , average increment in the band can be noticed. Figure 7.23 shows psd of original band and then modified psd is overlapped on the same band also. In the simulation result second graph shows clear plot of difference in the sixth band for reference. Thus, very accurate amount of sound intensity increment can be observed my taking appropriate multiplication in the MRA wavelet domain.
7.2.7 Seventh Band Enhancement

Figure 7.24 Original and reconstructed speech seventh band

Original speech signal

Modified speech signal

Figure 7.25 Original and modified wavelet coefficient seventh band

Wavelet coefficients

Modified Wavelet coefficient
Wavelet multi resolution tool is very useful for frequency band partition. Figure 7.24 shows simulation results of original speech and enhanced speech. In the developed research 8 level wavelet transform with wavelet db4 is used. Seventh band wavelet coefficients are selected and enhancement is performed between wavelet transform and inverse wavelet transform. Seventh band wavelet coefficients are multiplied with factor 1.3. Figure 7.25 shows simulation plot for two category of wavelet coefficient one original coefficients and other enhanced band coefficients of wavelet transform. When signal reconstructs after enhancement some magnitude variation is noticed. As per requirement any level of magnitude increment can be achieved. In present work for example 3db variation is taken. By multiplying with suitable numerical value in between with wavelet coefficient , average increment in the band can be noticed. Figure 7.26 shows psd of original band and then modified psd is overlapped on the same band also. In the simulation result second graph shows clear plot of difference in the Seventh band for reference. Thus, very accurate amount of sound intensity increment can be observed my taking appropriate multiplication in wavelet domain.
7.2.8 Eighth Band Enhancement

![Original speech signal](image1)

![Modified speech signal](image2)

**Figure 7.27** Original and reconstructed speech for eight band

![Wavelet coefficients](image3)

![Modified Wavelet coefficient](image4)

**Figure 7.28** Original and modified wavelet coefficient for eight band
Wavelet multi resolution tool is very useful for frequency band partition. Figure 7.27 shows simulation results of original speech and enhanced speech. In the developed research 8 level wavelet transform is used so different nine bands are generated. Eighth band wavelet coefficients are selected and enhancement is performed between wavelet transform and inverse wavelet transform. Eighth band wavelet coefficients are multiplied with factor 1.25. Figure 7.28 shows simulation plot for two category of wavelet coefficient one original coefficients and other enhanced band coefficients of wavelet transform. When signal reconstructs after enhancement some magnitude variation is noticed. As per requirement any level of magnitude increment can be achieved. In present work for example 3db variation is taken. By multiplying with suitable numerical value in between with wavelet coefficient , average increment in the band can be noticed. Figure 7.29 shows psd of original band and then modified psd is overlapped on the same band also. In the simulation result second graph shows clear plot of difference in the eighth band for reference. Thus, very accurate amount of sound intensity increment can be observed my taking appropriate multiplication in the wavelet domain.
7.2.9 Ninth Band Enhancement

Figure 7.30 Original and Reconstructed speech ninth band

Figure 7.31 Original and modified wavelet coefficient ninth band
In Figure 7.30 original and modified speech is shown. First simulation result shows original noise removed speech which is output of any adaptive algorithm. Either it may be LMS, NLMS or RLS. Then taken speech is required to modify as per the audiogram. First layout shows original and second shows band enhanced layout.

Second simulation result in Figure 7.31 two layouts can be observed. In that first output shows original coefficient after applying wavelet transform of eight stages. All the band coefficient are shown. In the second layout for each and every band modified coefficient are shown in the result after multiplication of some numerical value for the increment of 3 dB.

Third result in Figure 7.32 shows plot of original psd of signal. Original psd shows energy of natural band which is output of adaptive filter. Overlapping graph shows band enhancement after multiplying of some numerical coefficient. It can be observed that only some part of original psd plot has been changed. That reflects effect of wavelet coefficient multiplication effect in frequency domain easily. Moreover second plot shows clear difference between two plots in reality.