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Bandwidth reduction schemes for baseband control signal in DLM transmitter architecture

Master of Science Thesis

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Abstract

When dynamic load modulation (DLM) based transmitter architecture is used with broadband signals such as Wideband code division multiple access (WCDMA) or Long term evolution (LTE), the envelope signal required to control the load impedance and thus the RF output of PA has very wide bandwidth. It is very difficult to design a transmitter system for such a wideband control signal. In this thesis work, different schemes are investigated for the bandwidth reduction of wideband control signals in Dynamic load modulation (DLM) transmitter architectures.

The schemes are implemented using different techniques, based on digital signal processing (DSP) and the simulated results are shown in Matlab. These techniques are capable to reduce the bandwidth of the envelope signal while maintaining high average efficiency of power amplifiers.

A single carrier WCDMA signal having bandwidth 3.84 MHz is used as the desired output signal in demonstrating the techniques although the techniques can be well applied to multicarrier signals. The bandwidth of the optimal baseband envelope signal is 12.5 MHz with overall average efficiency of 54%. It is effectively reduced to 4 MHz while retaining high efficiency which is around 49%. The target reduced bandwidth is chosen as 4 MHz to show the effectiveness of the techniques, further in the condition of keeping relatively high average efficiency of PA. Moreover, the efficiency is also compared at different reduced bandwidths for the implemented methods.

ACKNOWLEDGMENT

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Chapter 1 Introduction

In today's third and fourth generation networks, the main issue with any base station is of relatively high power consumption. One reason for this is the complexity of base station due to extensive signal processing. However, the major source which consumes a lot of power in any base station is the power amplifier (PA). Power consumption in PA is due to the low mean efficiency where the majority of the input power is dissipated as heat. Therefore high efficiency is the primary concern in any RF PA. Thus PA plays a very important role to build an efficient transmitter architecture in modern wireless communication systems.

The efficiency is defined as a ratio of the generated RF power and the drawn DC power.

$$\eta = \frac{P_{\mathrm{out,RF}}}{P_{\mathrm{in,DC}}} \quad (1.1)$$

where $P_{\mathrm{out,RF}}$ is the output RF power, $P_{\mathrm{in,DC}}$ is the applied DC power and η represents the efficiency.

Another metric of interest, power added efficiency (PAE) is also used to measure the performance of PA and is defined as the ratio of the difference between the output power P_{out} and the input power P_{in} to the dc supply power P_{dc}.

$$\mathrm{PAE} = \frac{P_{\mathrm{out}} - P_{\mathrm{in}}}{P_{\mathrm{dc}}} \quad (1.2)$$

High efficiency is required for low energy consumption, a longer battery lifetime and thermal management. The efficiency of conventional linear RF PAs varies with the signal amplitude (envelope), resulting in relatively low average efficiencies, especially when the peak-to-average ratio (PAR), which is defined as the ratio of the peak power to the time-averaged power of the signal, is high.

$$C = \frac{P_{\mathrm{peak}}}{P_{\mathrm{avg}}} \quad (1.3)$$

where P_{peak} is the peak power, P_{avg} is the time-averaged power of the signal and C is the peak to average ratio. For example, for a Rayleigh-envelope signal with a 10-dB peak-to-average ratio, the average efficiencies of ideal class-A and –B PAs are only 5 and 28 percent, respectively [1].

Another very important property which defines the performance of any PA is the linearity. Linearity is required for achieving less distortion in amplified output signals and also to minimize interference and spectral re-growth. Many modern applications that use shaped-pulse data modulation or simultaneously transmit multiple carriers require linear RF power amplification. As an example, efficient modulation schemes in which the information is modulated in both the amplitude and phase result in time varying amplitude. In order to further increase the bit rate, the numbers of carriers need to be increased. This is possible by widening the bandwidth or by multiband solutions. This altogether results in high linearity requirements for PAs. Usually, linear class A or class AB PAs are used to achieve such high linearity, which, unfortunately, yields low average efficiency.

Therefore, the design of linear and efficient radio frequency PAs presents one of the most challenging design problems in modern telecommunication systems when using spectral efficient modulation schemes such as WCDMA, OFDM and new standards such as LTE.

The efficiency and linearity of PAs can be increased in many ways. The architecture of PA can be classified into two categories. The linear mode amplifiers and switch mode amplifiers.

Linear amplifiers can be combined in many ways. Recently various techniques for high-efficiency linear amplification (e.g. Chireix and Doherty [1]) have been developed, but all are subject to limitations in bandwidth or the dynamic range over which the efficiency is improved. In Doherty two or more amplifiers are connected in a parallel like configuration with a main and peak amplifier. By the proper combining of the amplifier branches the impedance seen by each amplifier for different power levels is beneficial to efficiency, compared to a single transistor amplifier.

Alternatively, the DC supply voltage of PA can be changed to keep the transistor working closer to saturation for different output levels. The technique used in Envelope Tracking (ET) is based on this principle. In ET, the DC supply voltage follows the envelope of the signal.

In order to reach very high efficiency figures, switch mode amplifiers can be used such as envelope elimination and restoration (EER). In EER, the maximum output power is changed by adjusting the drain DC voltage, while the transistor still kept working in saturation. But, EER has several short comings such as phase distortion introduced by non-linear transistors and by limiters.

The output power of an amplifier depends on the output swing at the amplifier output and also on the impedance seen by the amplifier. So by dynamically changing the load impedance, the output power is varied and this technique is known as dynamic load modulation (DLM).

This work is performed with a DLM transmitter architecture, which has been proven to be a promising technique in improving the average efficiency of PA as shown in [1] and [2]. In a DLM transmitter, high efficiency can be achieved in PA for a desired output signal by controlling the input signal amplitude and load impedance. The PAE of power amplifier in back off operation using DLM transmitters can significantly be enhanced by utilizing an optimum controlling scheme.

In DLM, a baseband control signal is required to change the load impedance. However, due to wider bandwidth requirements for 4^{th} generation networks, the bandwidth of the baseband control signal becomes very wide and is at least twice as that of RF input signal. It is very challenging to design a transmitter system for such a wideband control signal.

The bandwidth of the control signal should be reduced in order to make an efficient transmitter design. If the bandwidth of the control signal is reduced directly by passing it through the low-pass filter, the overall efficiency of PA is degraded and significant distortion is produced at the output. In this thesis work, several schemes are investigated for the bandwidth reduction of the baseband control signal while maintaining high efficiency and linearity.

Organization of thesis

The thesis report is organized into four chapters:

Chapter 2 contains an overview and background of DLM transmitter. First, different classes of conventional linear amplifiers are discussed. Also the limitations and drawbacks associated with these linear amplifiers are mentioned. Then, some concepts like load line theory, load modulation etc are explained and design of DLM transmitters are discussed in detail. The design of matching network, selection of impedance locus is also explained in this chapter. Then, a joint controlling scheme for the DLM transmitter is presented by designing an optimal RF input signal and an optimal baseband control signal. The results of simulations are also shown for the suggested controlling scheme.

In chapter 3, the motivation and principles for bandwidth reduction of baseband control signal are introduced and different techniques implemented for the bandwidth reduction are discussed in detail along with the achieved simulated results and efficiency curves.

In chapter 4, the comparison between the suggested techniques is made. In the last chapter, the thesis is concluded with possible future work.

Chapter 2 Dynamic Load Modulation Transmitter

Classes of Power Amplifiers

PAs are identified by their classes of operation. When used as dependent current source, amplifiers can be distinguished into several classes such as Classes A, B and C depending upon the conduction angle 2θ of the drain current i_D. The waveform for the drain current i_D of PA working as dependent current source is shown in Fig. 1.

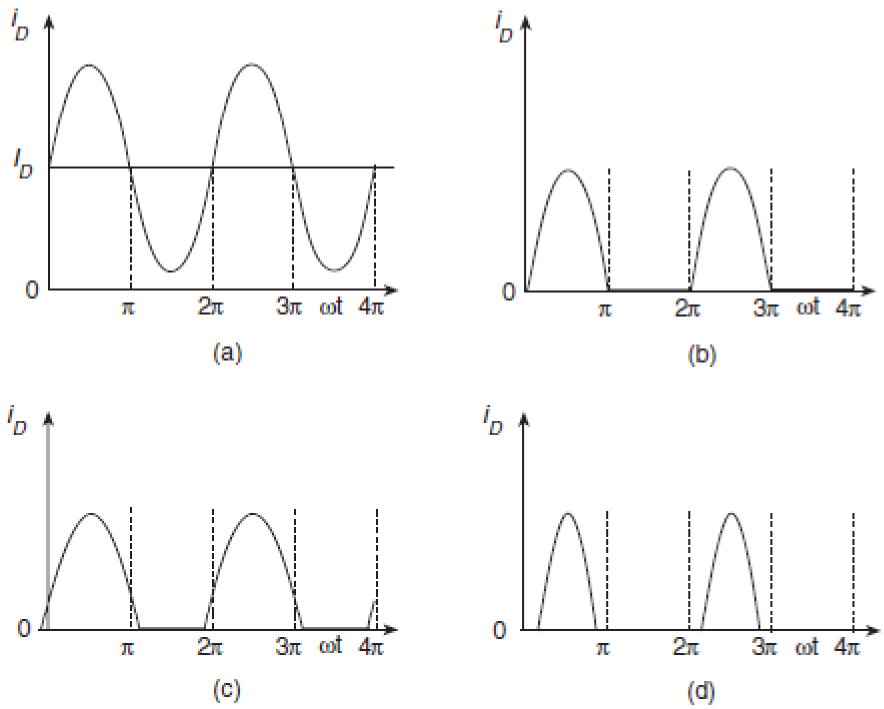

Figure 1: Waveform of current i_D in various classes of PA (a) Class A (b) Class B (c) Class AB (d) Class C

In Class A, the transistor is operating in active region all the times and acts as a current source which is controlled by the input signal. The conduction angle 2θ in Class A operation is 360° i.e. it conducts for the entire cycle. In the ideal case, Class A offers a maximum efficiency of 50%.

In Class B, the transistor is biased at the edge of cut-off, so in the absence of input signal there is no power dissipation (unlike Class A). Once the signal is applied, the transistor conducts for half of the input cycle (a conduction angle of 180°) and acts as a current source. The maximum efficiency under ideal conditions is 78.5%.

In Class AB, the transistor is biased slightly above cut-off. In this case, the conduction angle 2θ is between 180° and 360°, and the term Class AB is used for the amplifier.

In Class C, the transistor is biased in the cut-off region and for only a portion of input signal, which is less than 180°, acts as a current source. Assuming a sine-wave input, the output current is tips of a sine-wave.

The operating point for various classes of operation is shown in Fig. 2.

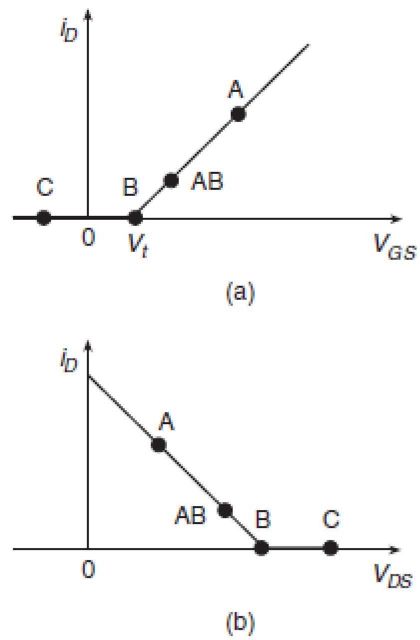

Figure 2: Operating points for various Classes of PA

Condition for maximum efficiency

The average drain efficiency of PA as described in equation (1.1) can also be written as:

$$\eta = \frac{P_{\mathrm{out,RF}}}{P_{\mathrm{in,DC}}} = \frac{P_{\mathrm{DS}}}{P_{\mathrm{in,DC}}} = 1 - \frac{P_{\mathrm{D}}}{P_{\mathrm{in,DC}}} \qquad (2.1)$$

The condition for achieving 100% average efficiency is:

$$P_{\mathrm{D}} = \frac{1}{T}\int_0^T i_D V_{DS}\, dt = 0 \qquad (2.2)$$

or

$$i_D V_{DS} = 0 \qquad (2.3)$$

Thus the waveform for i_D and V_{DS} should be non-overlapping for achieving an efficiency of 100%, at-least in theory.

Such non-overlapping waveforms are possible in switching amplifiers classified as D and E. In these Classes, the transistor acts as a switch and not a current source. Since an ideal switch has either zero voltage across it or zero current though it, the switch dissipates no power. Therefore, switching classes of amplification have an efficiency of 100% in the ideal case.

The Class D amplifier is similar to an inverter gate with a series tuned circuit connecting its output to the load. The input signal toggles the output of the gate and generates a square waveform which is filtered by the tuned circuit, and a sinusoidal waveform appears across the load. The on-resistance of the transistors reduces the efficiency from its ideal 100%.

Class E amplifiers have a single switching transistor connected to a passive load network. Because Class E uses capacitance shunting across the switch to shape the voltage and current waveforms, it avoids the power loss due to charging and discharging the capacitance, thus, achieving a better efficiency than Class D at high frequency. Class E is also less sensitive to the transition time of the switch than Class D.

However, in modern modulation techniques such as WCDMA and OFDM, the peak to average ratio is very high and thus PA has to be operated in back-off state to avoid non-linearity and signal distortion. But by backing off the PA, the efficiency is considerably reduced. It is shown for the first time in [3] that by using DLM transmitter architecture under back off, the PAE of the PA is improved significantly.

Load Line

The maximum output power from any transistor can be obtained if the load impedance is matched to the transistor. Therefore load line match is used to serve this purpose. The maximum drain current of any transistor $I_{\max}$ depends on the transistor technology and size of the die. This maximum current handle by the transistor is the limiting factor for the peak output power. Similarly, the maximum voltage supplied to the drain V_{DC} is limited by the available power supply or the break down voltage of the transistor [4]. The full swing of the drain voltage is 0 to $2V_{\mathrm{DC}}$ and of the drain current is 0 to $I_{\max}$. The optimal location of the operating point should be the middle of the load line. So the optimal load impedance is:

$$R_{\mathrm{opt}} = \frac{V_{\mathrm{DC}}}{\frac{I_{\max}}{2}} \qquad (2.4)$$

Load Modulation- Principle

Load modulation is a technique where the drain efficiency can be improved by changing the effective load impedance seen by the transistor. The effective change in load impedance must be on the intrinsic drain of the transistor, since this is the point where the drain voltage drives a current into the output network and thus, where the power is generated. A change of impedance outside of the intrinsic drain of the transistor may give unexpected results [4]. The basic concept of the load modulation is to dynamically change the load line match depending on the instant demand of the output power.

DLM Transmitter

In DLM, a tunable matching network is used to control the output signal. The impedance of tuning network is varied by applying an external control signal which in turn varies the output signal.

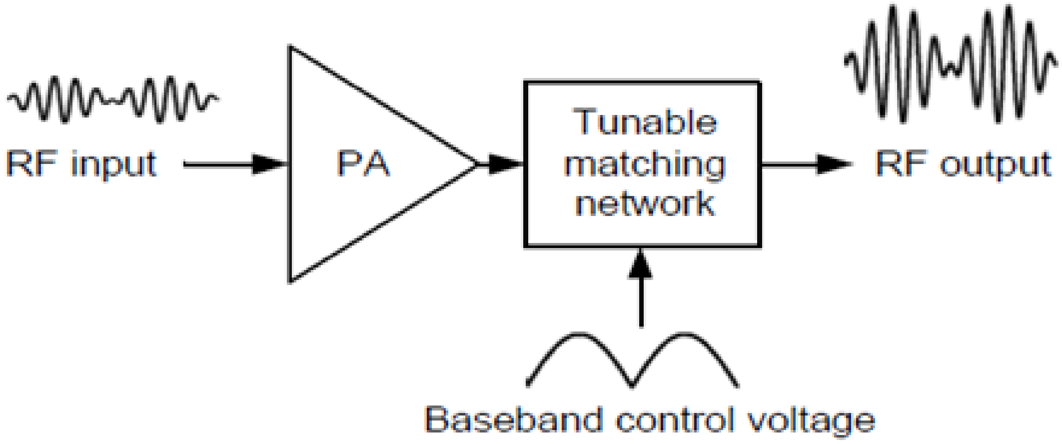

Figure 3: Dynamic load modulation transmitter architecture [3]

An electronically tuned RF PA can be used to realize such a network. Time-varying bias or control voltage is applied to the electronically tunable matching network, which in turn varies the drain-load impedance, resulting in time-varying signal amplitude at the output [1].

The control voltage applied to the matching network varies the instantaneous load impedance presented to the RF PA along a locus. High efficiency and dynamic ranges can be achieved by careful choice of the impedance locus. The ideal locus depends upon the type of PA, and load-pull contours of output power and efficiency are useful in determining the preferred locus [1].

As an example, the power and efficiency "load-pull" characteristics of an ideal class-E PA are shown in Fig. 4 [5]. The output power decreases along the locus of ideal class –E PA denoted by circles, as the impedance becomes more reactive. The efficiency is 100% for the straight line which runs through the centre at an inclination of 65° and it decreases as the locus moves away from this line. Therefore the ideal LM line starts from the centre where the load is pure resistive and it moves over this 100% efficiency line to the edge of the chart. Such a locus maintains 100% efficiency for all the output powers.

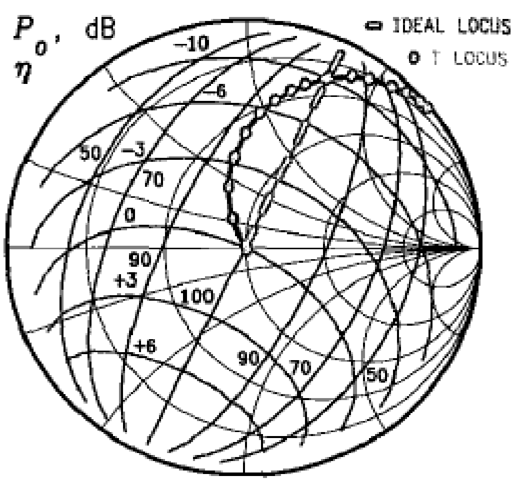

Figure 4: Load Pull characteristics of Class -E amplifier [5]

Matching Network

Various configurations can be chosen for the tunable matching network to achieve the required impedance tuning for efficiency enhancement. However, not every configuration is appropriate when there are constraints like linearity, peak output power levels and high frequency. T-networks with variable capacitors and resistors are used in matching network. As can be seen in Fig. 5, a simple T-network is employed with single variable element and a considerable improvement in efficiency is achieved at lower output amplitudes [1].

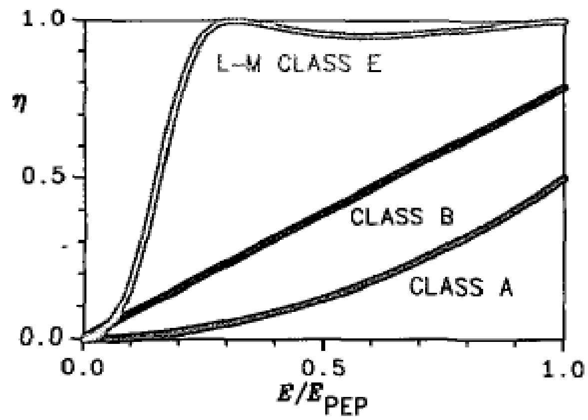

Figure 5: Efficiency comparison of different PA Classes [5]

However, in case of high PAR signals, the PA works in back- off conditions most of the time, which significantly lowers the efficiency. Thus efficiency boost is required, which is only possible through a careful choice of matching network.

In [3], a varactor based tunable matching network is implemented by using abrupt junction silicon varactors and a significant improvement in efficiency is achieved at 1 GHz frequency with peak output power level of 7W. The results are compared with the same PA having fixed impedance of 50Ω as shown in Fig. 6.

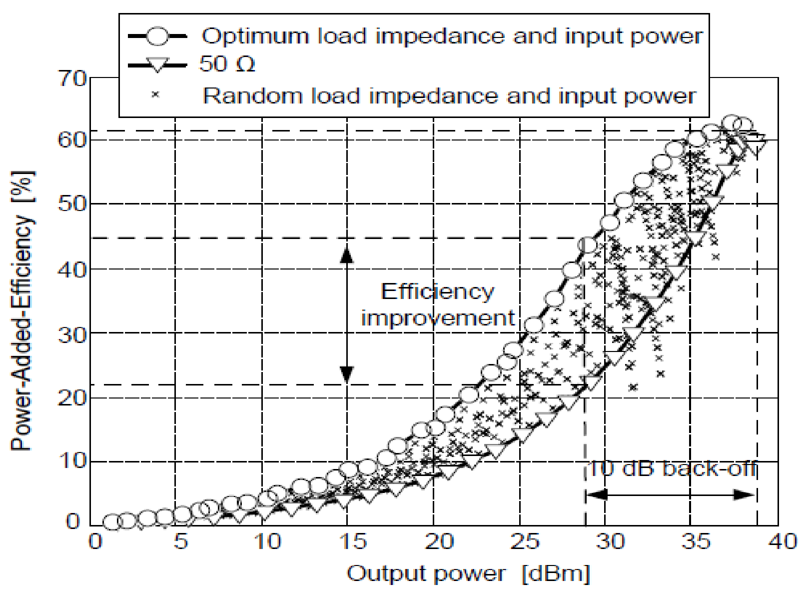

Figure 6: Load-pull measurements of a 1 GHz LDMOS class-E PA [3]

There are also some issues when using varactor based matching networks. Like, for high power applications, varactors with high break-down voltages and low series resistance are required. Besides, the achievable dynamic range of output power is also limited as it is directly related to the available tuning range of the varactor. Linearity of the varactors is also a major issue. Latter two of the issues can be handled through utilizing an efficient controlling scheme.

That is, an efficient and optimized controlling scheme is needed to enhance the efficiency of PA over a wide dynamic range and with high linearity. In the next section, a general control scheme is presented which is based on a joint control of by optimal input RF signal and an optimal control signal.

Optimal Controlling Scheme

The desired RF output signal can be produced by co-controlling the RF input signal and the baseband control signal. The relationship between different variables can be expressed as [2]:

$$V_{\mathrm{out,RF}} = f_1(V_{\mathrm{in,RF}}, V_{\mathrm{cc}}) \qquad (2.5)$$

where the RF output signal, the RF input signal and the baseband control signals are represented by $V_{\mathrm{out,RF}}$, $V_{\mathrm{in,RF}}$ and V_{cc} respectively. The function f_1 describes the relationship between the variables. Memory effects are not considered in equation (2.5) and the function f_1 only depends on PA characteristics.

However, the relationship of $V_{\mathrm{out,Rf}}$ with $V_{\mathrm{in,RF}}$ and V_{cc} is not one to one, i.e. several combinations of $V_{\mathrm{in,RF}}$ and V_{cc} are possible that gives the same $V_{\mathrm{out,Rf.}}$ However, the efficiency and power consumption will be different for each combination. It is shown in [2], that by carefully observing the input variables, a combination of V_{cc} and V_{in} can be selected which minimizes the consumption power in PA, and thus maximizing the overall efficiency of the transmitter system.

Furthermore, each selected combinations corresponding to different $V_{\mathrm{out,RF}}$, should follow a path on the Smith chart that is possible to practically realize using an electronically tunable matching network [2].

Using the above conditions, it is possible to find the following efficiency optimized parameters:

$$V_{\mathrm{in}} = f_{in.opt}(V_{\mathrm{out}}) \qquad (2.6)$$

and similarly,

$$V_{\mathrm{cc}} = f_{c,opt}(V_{\mathrm{out}}) \qquad (2.7)$$

Where $f_{in.opt}$ and $f_{c,opt}$ represents the optimized functions for the RF input signal and the baseband control signal. It is possible to calculate the required controlling signals i.e. V_{in} and V_{cc} by using these functions, if the output signal V_{out} is known.

The surface showing the relationship between these variables is plotted in Fig. 7. The data is calculated from the static measurements on DLM transmitter. The optimal path is also indicated in green in Fig .7.

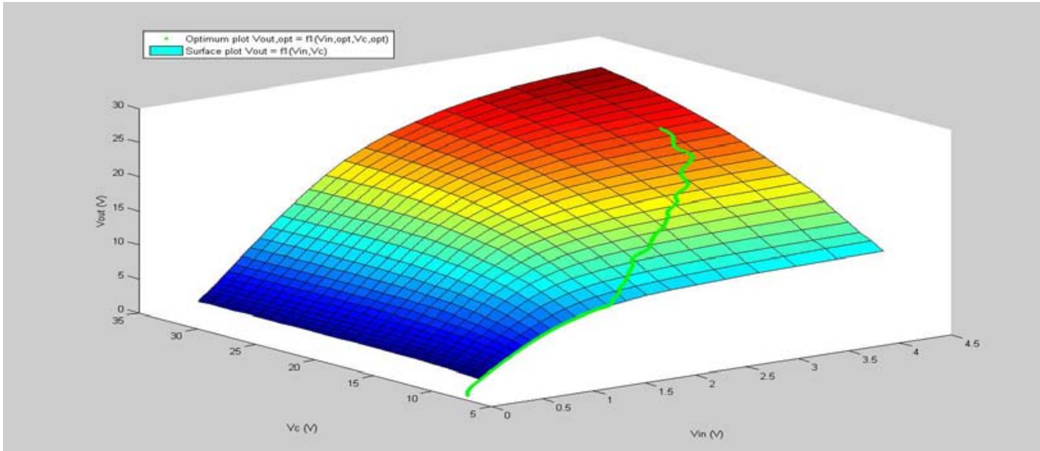

Figure 7: Surface plot showing the relationship of Vin, Vcc and Vout. Green line shows the optimal path.

DLM with Optimal Control Signal

In this section, the performance and working of DLM transmitter is shown in ideal conditions by using the envelope optimal RF input signal $V_{\mathrm{in,opt}}$ and the optimal baseband control signal $V_{\mathrm{cc,opt}}$,to get the envelope of the desired output i.e. target V_{out} .

The optimized functions $f_{in.opt}$ and $f_{c,opt}$ from equations (2.6) and (2.7) are used to generate $V_{\mathrm{in,opt}}$ and $V_{\mathrm{cc,opt}}$ respectively from the given target V_{out}. The set of $V_{\mathrm{in,opt}}$ and $V_{\mathrm{cc,opt}}$ is then applied to PA+MN block to get the optimal output $V_{\mathrm{out,opt}}$ which is the exact replica of target V_{out} as can be seen in Fig. 8.

By using the optimal conditions, although high average efficiency and linearity are achieved but the bandwidth of control signal becomes very wide. The spectrum of the optimal control signal and the RF output signal is also compared at the end of this section to enlighten the need for bandwidth reduction. In next chapter, the techniques for bandwidth reduction are discussed and the results of simulations are shown.

A single carrier WCDMA signal is used as target output in simulations. The bandwidth of target output signal is 3.82 MHz at 99.99% energy. The single carrier WCDMA signal is chosen due to the limitation of test bench, although this technique can also be used with multi carrier signal.

The block diagram representation of such a system is shown in Fig. 8:

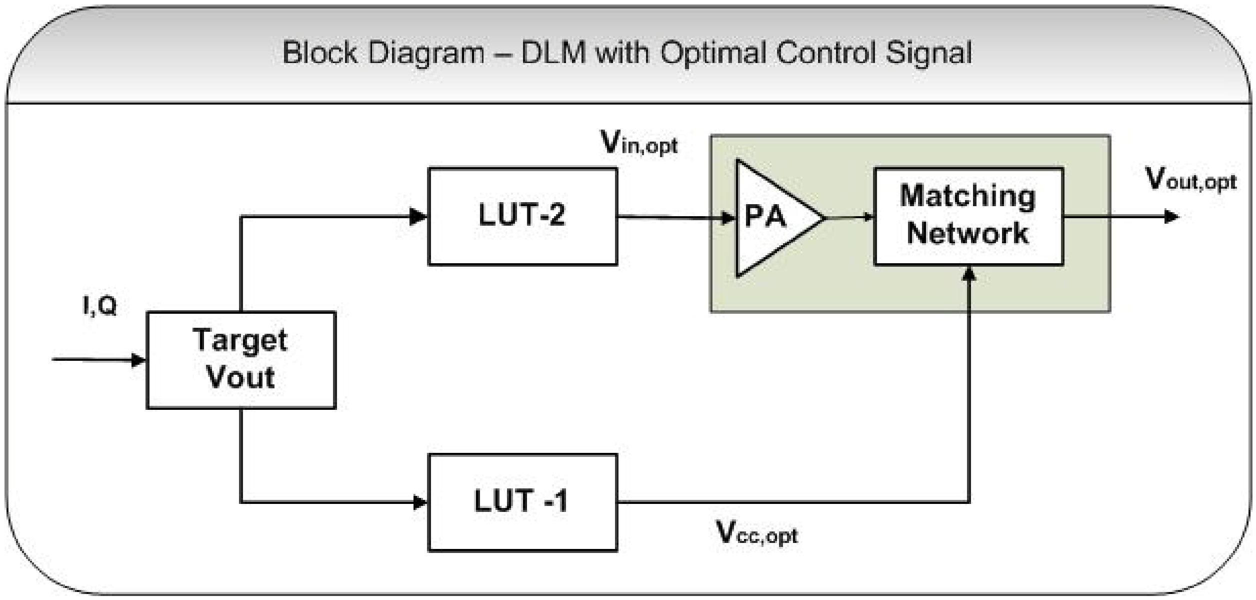

Figure 8: Block diagram of DLM with optimal control signal

The PA and the matching network sub-blocks are cascaded in a single block and implemented in form of a look-up table which looks for the value of V_{out} based on the values of V_{cc} and V_{in}. It is the realization of equation (2.5) and the data is obtained through static measurements.

Target V_{out} is the single carrier WCDMA signal. This target output is passed through the lookup table blocks LUT-1 and LUT-2 to produce $V_{\mathrm{cc,opt}}$ and $V_{in,\mathrm{opt}}$. The LUT-1 and LUT-2 blocks as described above are the realization of the optimized functions $f_{in.opt}$ and $f_{c,opt}$. These functions are constructed from the optimal control variables of RF input and baseband control signals by using polynomial curve fitting. The optimal variables are obtained through static measurements.

Polynomial Curve Fitting

When many sample data pairs $\{(x_k, y_k), k = 0{:}\, M\}$ are available, it is often required to calculate the relationship between the two variables or to describe the trend of the data, in a form of function $y = f\,(x)$. It is not needed to find a function passing exactly through every point. Instead of pursuing the exact matching at every data point, an approximate function is formulated in form of polynomial that describes the data points as a whole with the smallest error in some sense, which is called the curve fitting.

As a reasonable means, the least-squares (LS) approach is considered to minimize the sum of squared errors, where the error is described by the vertical distance to the curve from the data points. In Fig. 9, a continuous curve is fitted by using polynomial of degree 7 and using the LS approach.

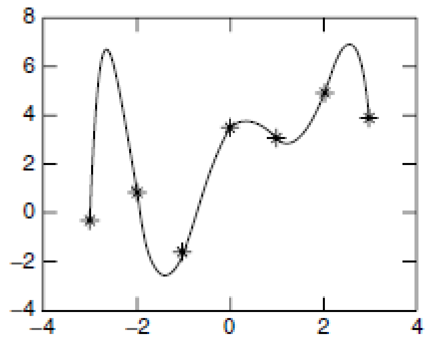

Figure 9: Polynomial curve fitting by the LS method

Optimal Signal Generating Functions

The relationship between the variables $V_{\text{out,opt}}$ and $V_{\text{in,opt}}$, $V_{\text{out,opt}}$ and $V_{\text{cc,opt}}$ are formulated by calculating the optimized functions $f_{in.opt}$ and $f_{c,opt}$ through polynomial curve fitting and these fits are then used to calculate the desired RF input and baseband control signals to achieve the target $V_{\text{out,opt}}$.

In Fig. 10, the optimized functions $f_{in.opt}$ and $f_{c,opt}$ obtained through curve fitting are plotted along with the actual measured data. These functions are then used in LUT-1 and LUT-2 blocks to generate the desired input signal and the baseband control signals from the given output signal.

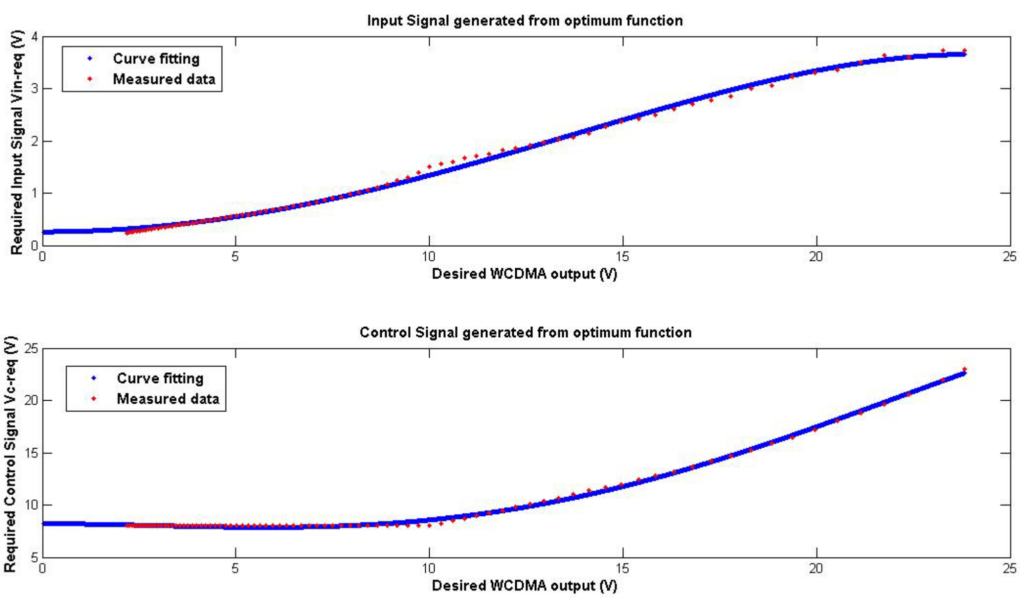

Figure 10: Plots of optimum control signal and optimum input signal generated from target output

The curves showing the relationship between the RF input and the baseband control signals with the RF output signal is shown in Fig. 10. A 5th order polynomial model, which includes only odd order, is used for the RF input signal. While the optimum control signal is modeled with polynomial order 4 with both even and odd orders. The set of $V_{\mathrm{cc,opt}}$ and $V_{\mathrm{in,opt}}$ is then applied to the PA and the matching network blocks.

PA and Matching Network Block

Static measurements were performed on the DLM transmitter (PA+MN) and the data is stored in a separate table for each of the two input signals and the output signal. Then a two-dimensional interpolation method is used to interpolate the value of RF output signal based on the values of RF input signal and the baseband control signal. The method uses cubic interpolation technique and is implemented by using the Matlab built-in function 'interp2'.

For example, if the data points for three variables X, Y and Z are stored in separate tables or matrices and a relationship is formulated between the variables as:

$$Z = f(X, Y) \quad (2.8)$$

Now, by using the two-dimensional interpolation, the output ZI can be calculated for any set of input data-points XI, YI.

The Matlab function ZI = interp2(X,Y,Z,XI,YI) returns matrix ZI containing elements corresponding to the elements of XI and YI and determined by interpolation within the two-dimensional function specified by matrices X, Y, and Z.

Thus, by applying the $V_{\mathrm{cc,opt}}$ and $V_{\mathrm{in,opt}}$ to PA and MN block, it gives $V_{\mathrm{out,opt}}$ through interpolation, which is exactly same as target V_{out}.

Therefore, the optimum set of control signal and RF input signal is used to attain high efficiency in the PA and to achieve the desired output with high linearity.

Simulation Plots

The signal plots for optimal baseband control signal, RF input signal and achieved output signals are shown in Fig. 11.

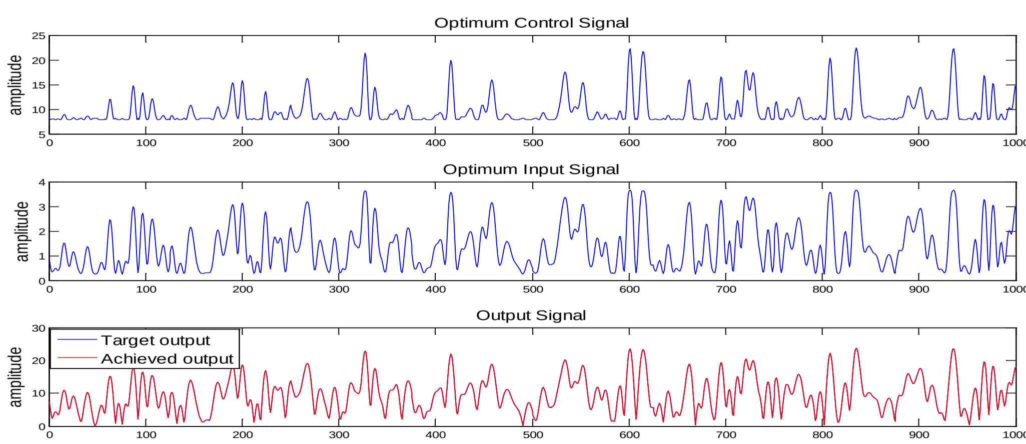

Figure 11: Signal plots of optimum control and input signal and comparison of achieved and target outputs

The average PAE achieved is 54.36% which is quite high. The spectrums of the optimal RF input, optimal control and RF output signals are shown in Fig. 12. Since the control signal is at baseband, therefore the spectrum is single sided whereas the input and output signals have double sided spectrum as they are at RF frequency.

The bandwidths are measured at 99.99% energy. The bandwidth of single carrier WCDMA target output signal is 3.84 MHz while the optimal control signal is very wideband and has a bandwidth of 12.5 MHz which is around 3.2 times as that of the target output signal. The RF input signal is also wideband and has a bandwidth of 10 MHz.

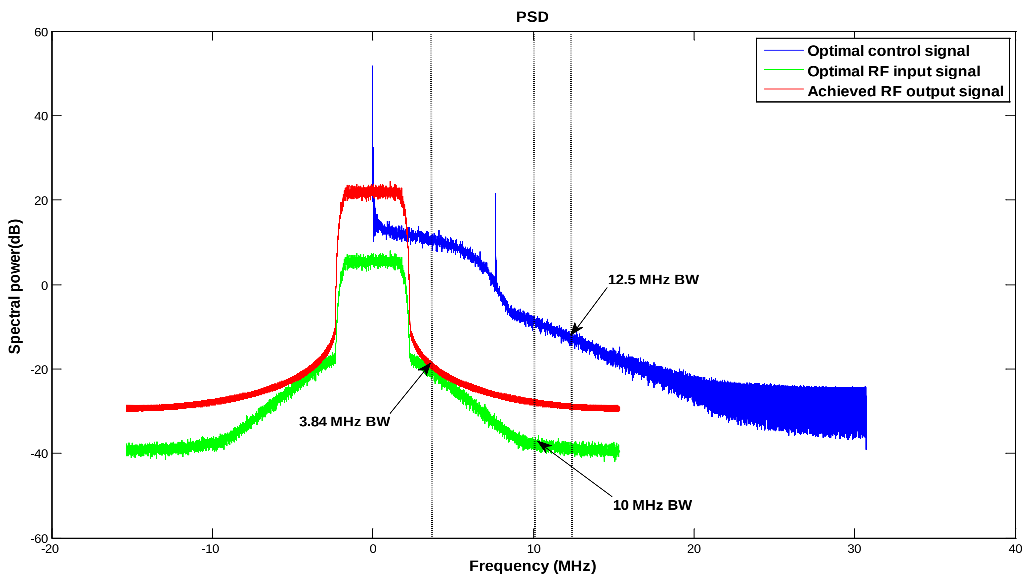

Figure 12: PSD plots for optimum control signal, RF input signal and RF output signal

The wideband envelope signal makes the transmitter design very difficult. The bandwidth of control signal should be reduced in order to simplify the transmitter design. In the next chapter, the principal of bandwidth reduction is discussed and the conditions necessary for reducing the bandwidth effectively and efficiently are explained.

Chapter 3 Bandwidth Reduction Schemes

Bandwidth Reduction Principle

The optimal control signal can be reduced in bandwidth under some conditions without much degradation in efficiency. It is shown here that a slow varying signal derived from the ideal optimal control signal can be used in DLM transmitters with overall high average efficiency of the system. The efficiency will not be as high as in case of ideal optimal control signal since the slow varying control signal may not follow the peaks of the RF input signal and thereby reducing the efficiency. The conditions needed for envelope signal are [6]:

- The amplitude of control signal should be large enough to avoid clipping of the output signal.
- The control signal should be conveyed with such accuracy that if it introduces any distortion in RF output signal, it can be corrected by varying the RF input signal.

The first condition is ensuring the peak power demands of the RF output signal while the second condition assures the linearity of the RF output signal.

The bandwidth can be reduced effectively under these two conditions while maintaining high efficiency. However, the bandwidth reduction also produces some distortion at the output which is corrected by pre-distorting the RF input.

The above two conditions can be written in form of a constraint equation which states that at any point the amplitude of reduced bandwidth signal should be equal or greater than that of the optimal control signal [6]:

$$V_{\mathrm{cc,red}} \geq V_{\mathrm{cc,opt}} \qquad (3.1)$$

Equation (3.1) is sufficient to maintain the conditions for bandwidth reduction as stated above. Hence, the effective reduced bandwidth signal is obtained by passing the optimal control signal through the low-pass filter and ensuring the equation (3.1) .The first condition is clearly satisfied in equation (3.1), i.e. the peak power demands of the RF output signal can only be well served if $V_{\mathrm{cc,red}}$ has equal or greater amplitude than that of $V_{\mathrm{cc,opt}}$. It is also fulfilling the second condition as can be seen in Fig. 2, any value of the RF output signal pointed by $V_{\mathrm{cc,opt}}$ and $V_{\mathrm{in,opt}}$ can be regenerated with $V_{\mathrm{cc,red}}$ and pre-distorting the RF input signal $V_{\mathrm{in,PD}}$ if, $V_{\mathrm{cc,red}} \geq V_{\mathrm{cc,opt}}$.

The bandwidth can be reduced arbitrarily with overall high average efficiency and linearity of the RF output signal by constructing the new RF input signal according to the bandwidth reduction baseband control signal, as long as equation (3.1) is satisfied.

Bandwidth Reduction Schemes

Two different schemes are studied and investigated for the bandwidth reduction of the baseband control signal and are implemented in Matlab. Firstly, an overview of each technique is discussed. Then the block diagram is presented and each block is explained in detail. Finally, signal plots, spectrum plots and the achieved results are shown.

Max - Filter based Method

Max-Filter based technique is very promising and is fairly simple. It is discussed and implemented in [7] for envelope tracking transmitter architecture and the results obtained are very impressive. To be used in DLM transmitters, the design is modified and some new blocks are inserted.

Block Diagram

The block diagram of the bandwidth reduction scheme used in max value filter based method is shown in Fig. 14:

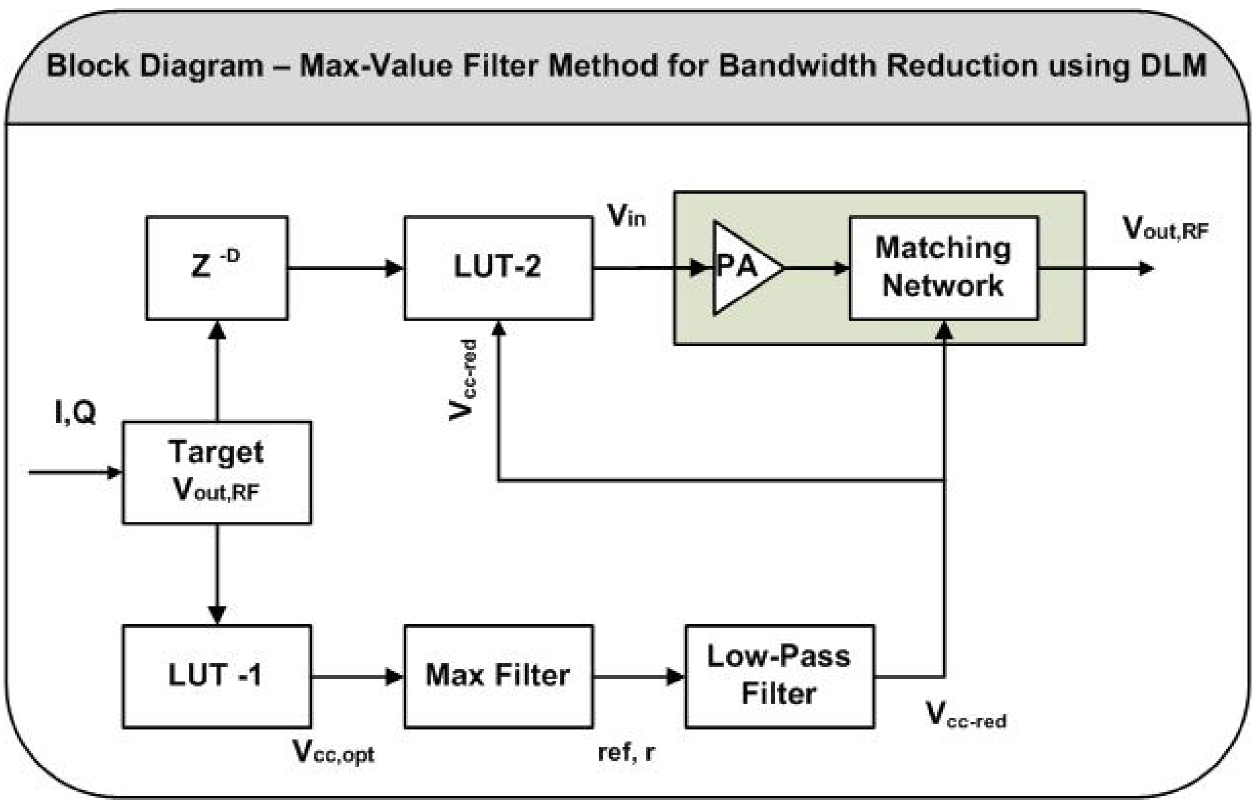

Figure 13: Block diagram for max value filter method

In this block diagram, compared to Fig. 3 which is the ideal case, three new blocks are inserted which are Shift Register Comparator block (SHR), Low-pass Filter block and delay block.

Maximum Value Filter

The purpose of maximum value filter is to generate a reference signal out of optimal control signal, which should meet the peak power demands of RF the output signal.

The optimal control signal has wide bandwidth and when it is passed through a LPF, the resultant filtered signal will be slowly varying and will be lower in amplitude than the original signal. Thus equation (3.1) will not be satisfied which will cause distortion and clipping at the output. So to meet the peak power demands and to prevent excessive distortions at the output, it is necessary to fulfill the condition stated in equation (3.1) by using some technique. Here maximum value filter is used to serve this purpose. Mathematically,

$$r[\,n] \,=\, max\{\mathrm{V}_{\mathrm{cc,opt}}\,[\,\mathrm{n}], \mathrm{V}_{\mathrm{cc,opt}}\,[\,\mathrm{n}\,-\,1], \ldots\ldots, \mathrm{V}_{\mathrm{cc,opt}}\,[\,n\,-\,D]\}$$

It generates a reference signal from the optimal control signal by comparing the current sample with "D" previous samples and replacing it with the maximum amplitude sample. Therefore, the amplitude of reference signal is constant for every D samples and is equal to the largest amplitude within those D-samples. The order "D" is proportional to the length and cutoff of the low-pass filter. The reference signal $r[\,n]$ serves the peak power demands of output signal by satisfying equation (3.1). It loosely follows the original optimal control signal as it is a kind of multilevel square waveform tracking the peaks of optimal control signal.

Low-Pass Filter (LPF)

The reference signal constructed by the SHR is applied to the low-pass filter which is implemented as a FIR filter of order 100. The purpose of LPF is to produce a slow varying reduced-bandwidth signal which should follow the crests of original optimal control signal as close as possible. FIR filter has linear phase response and constant group delay. The cutoff frequency of low-pass filter is varied from 0 to 1 (normalized Nyquist cutoff frequency) to see the effect of bandwidth reduction at each cutoff.

Delay block

The delay block is used to compensate for the delay introduced by the linear phase FIR filter. The original and filtered signals are correlated in order to calculate the delay between them. Then the target output signal is delayed by the same amount for delay compensation and is used to construct the RF input signal later.

PA and Matching Network Block

The PA and matching network (MN) blocks are cascaded into a single block and it produces linear amplified RF output signal based on the RF input signal and baseband control signal. MN is used to dynamically control the output signal amplitude. The block is implemented in form of a look up table and the data is obtained through static measurements.

Lookup Table -2 (LUT-2)

The function of LUT-2 block is to generate RF input signal based on the values of baseband control signal and RF output signal.

In chapter 2, the LUT-2 block is implemented as an optimal function generator which produces optimal RF input signal based on a given target output. However, after the bandwidth reduction of optimal control signal, significant distortion is introduced at the output and hence it is needed to adjust the RF input signal to achieve a non-distorted linear output. Therefore, the RF input signal is pre-distorted to cancel the distortions introduced by bandwidth reduction.

LUT-2 block is implemented as a two-dimensional look-up table. It searches for the value of pre-distorted RF input signal, in order to produce a linear output based on the values of target output and the reduced bandwidth control signal. It is obtained by inversing the lookup table implemented for PA and matching network blocks.

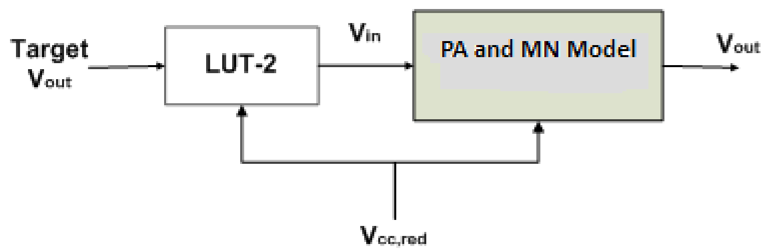

Figure 14: LUT-2 block with PA and MN block

As can be seen in Fig. 15, the LUT-2 block generates V_{in} based on target V_{out} and $V_{\text{cc,red}}$. This $V_{\text{cc,red}}$ along with the V_{in} obtained are then applied to PA and MN block to get V_{out} which is highly linear and is exactly same as target V_{out}.

Simulation Plots

In simulations, a WCDMA carrier is used as desired output signal having bandwidth of 3.84 MHz while the target reduced bandwidth of baseband control signal is 4 MHz in below signal and PSD plots. The efficiency achieved is also compared at different bandwidths.

The first 1000 samples of the different signals obtained at various stages of ‘max-value filter method’ are shown in Fig. 16.

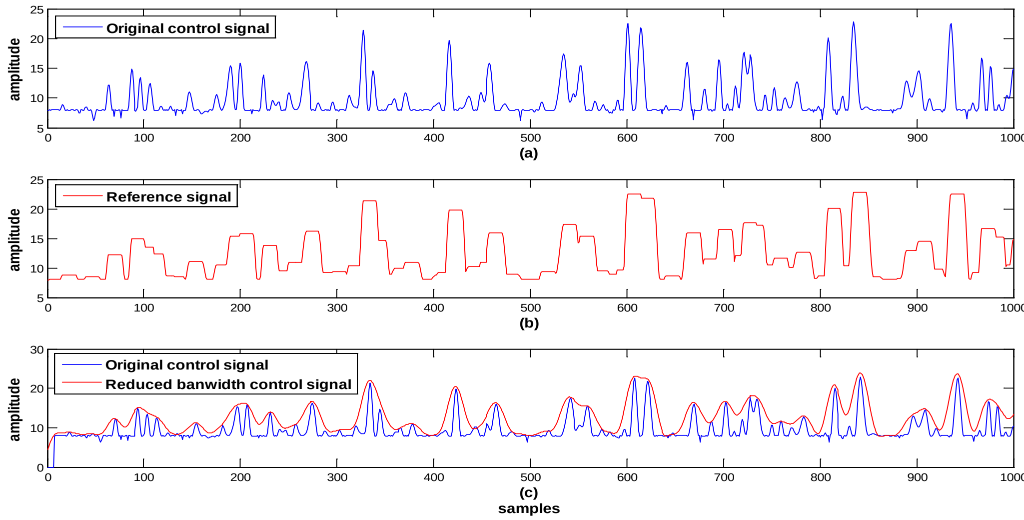

Figure 15: Signal plots at various stages of the method

- Fig. 16(a) is showing the original optimal control signal which has to be reduced in bandwidth.
- The reference signal after max-value filter is shown in Fig. 16(b). The reference signal is passing through the peaks and remains flat over some samples in proportional to the delay of low pass filter to ensure the peak power demands of the output signal.
- Fig. 16(c) shows the reduced bandwidth signal along with the original control signal. Reduced bandwidth signal is a slowly varying signal which tightly follows the crests and loosely follows the troughs of original control signal.

PSD plots for the optimal and reduced bandwidth baseband control signals are compared in Fig. 17. The bandwidths are marked at 99.99% energy points.

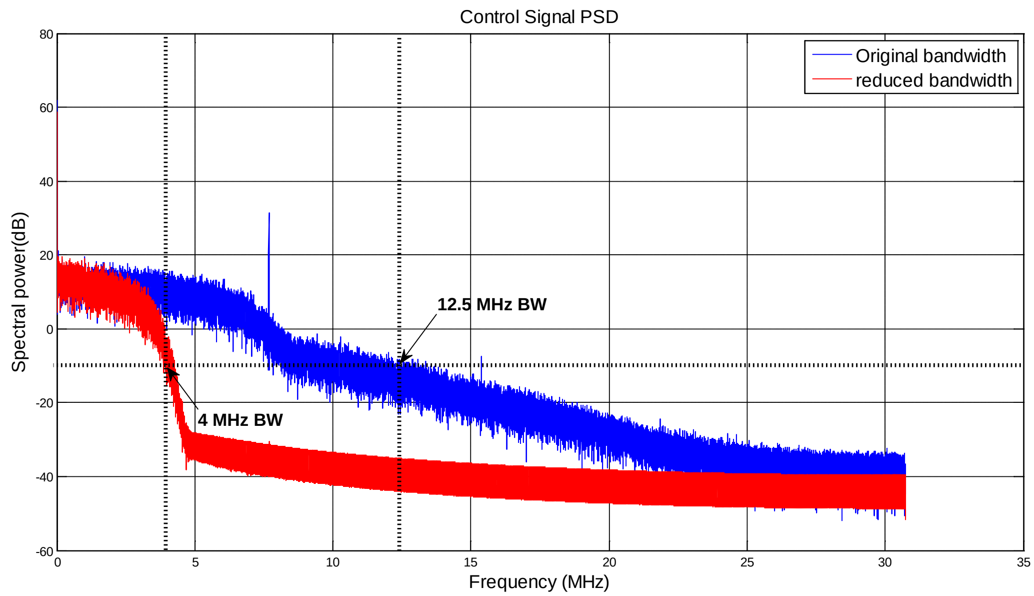

Figure 16: PSD plots for original and reduced bandwidth control signal and target and achieved output signals

Fig. 17 clearly shows the reduction in bandwidth of the baseband control signal while achieving highly linear output signal. The bandwidth is reduced to 4 MHz from 12.5 MHz by using this technique while the efficiency (PAE) achieved is 49% which is still very high.

The efficiency degradation is very less and is only around 5% as compared to the optimal efficiency whereas the reduction in bandwidth is highly significant i.e.it is reduced to 3 times. This shows the effectiveness of the technique which results in dramatic reduction of bandwidth while retaining high efficiency and linearity.

The efficiency is then measured at different bandwidths to show the degradation in efficiency at various bandwidths by using this technique and the curve generated is plotted in Fig. 17.

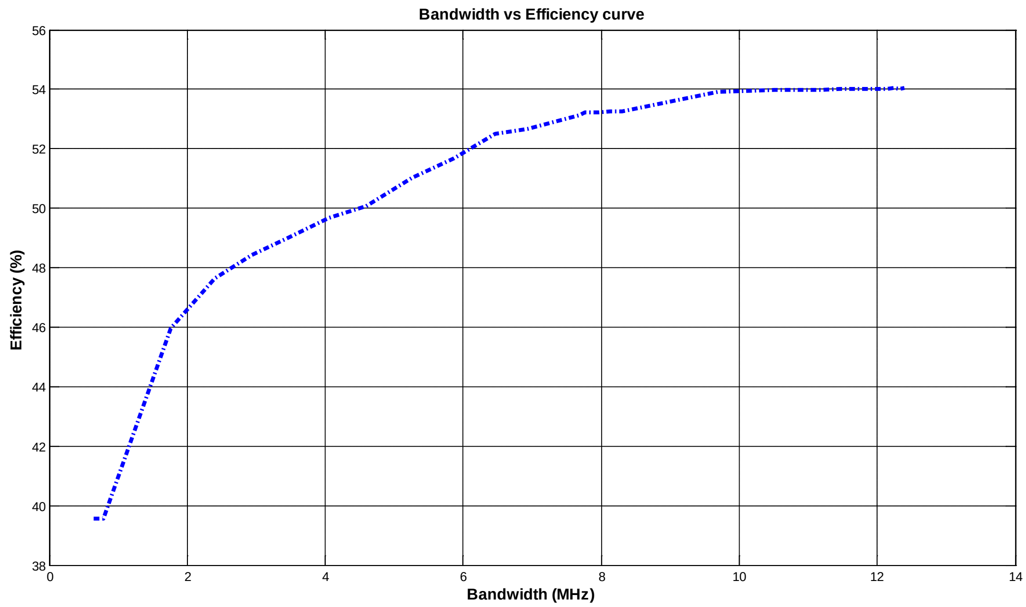

Figure 17: Bandwidth vs. Efficiency curve for 'Max-Filter Method'

It can be seen in Fig. 18, that efficiency is almost constant up to 10 MHz and is equal to the optimal efficiency i.e. 54% and then it reduces gradually up to 2 MHz at which the efficiency is 46.5%. Below 2 MHz, efficiency reduces drastically.

Dual Filtering Method

Overview and background

Dual filtering method is a completely different technique and it is producing very good results. The principle of the technique used in this method is:

The optimal baseband control signal $V_{\mathrm{cc,opt}}$ is passed through the low pass filter to get the filtered signal $V_{\mathrm{cc,filt}}$.

Let n represents the discrete time sample. Then, the difference signal $V_{\mathrm{d}}(n)$ is obtained by subtracting $V_{\mathrm{cc,filt}}(n)$ from $V_{\mathrm{cc,opt}}$(n). Mathematically,

$$V_{\mathrm{d}}(n) = V_{\mathrm{cc,opt}}(n) - V_{\mathrm{cc,filt}}(n) \quad (3.2)$$

The impulse response of the low pass having length L is represented by h_{LPF} . It is then normalized with respect to the maximum amplitude sample of the impulse response. The normalization is done in order to increase the amplitudes of impulse response which in turn reduce the number of iterations by producing filtered signal with increased amplitude as stated in equation (3.2). From the difference signal $V_{\mathrm{d}}(n)$, blocks of L samples from $n - L/2 \ to \ n + L/2$ are formed for all the values of n. Each block is then multiplied with h_{LPF} to get the resultant signal $V_{\mathrm{d,imp}}(n)$ as described in equation (3.2).

$$V_{\mathrm{d,imp}}[n-L/2,\ldots,n,\ldots,n+L/2] = V_{\mathrm{d}}[n-L/2,\ldots,n,\ldots,n+L/2] \times h_{\mathrm{LPF}}[0,\ldots,L-1] \qquad (3.3)$$

Instead of multiplying the difference signal with the impulse response of low –pass filter, the difference signal which is enrich in frequency content can also be passed through another low pass filter as shown in [6].

$V_{\mathrm{d,imp}}(n)$ is then added back to $V_{\mathrm{cc,filt}}$(n) to get the reduced bandwidth signal $V_{\mathrm{cc,red}}(n)$, which satisfies the conditions stated for bandwidth reduction in section 3.1.

$$V_{\mathrm{cc,red}}(n) = V_{\mathrm{d,imp}}(n) + V_{\mathrm{cc,filt}}(n) \qquad (3.4)$$

In equation (3.4), the difference signal $V_{\mathrm{d,imp}}(n)$ is added directly to the filtered signal $V_{\mathrm{cc,filt}}\ (n)$. This can cause an inverse effect at some locations. That is, if for any value of n:

$$V_{\mathrm{cc,filt}}(n) > V_{\mathrm{cc,opt}}(n)$$

Then, using equations (3.2) & (3.3),

$$V_{\mathrm{d}}(n),\ V_{\mathrm{d,imp}}(n) < 0$$

Therefore, in equation (3.4), the amplitude of the resultant signal i.e. $V_{\mathrm{cc,red}}(n)$ gets smaller for that value of n.

Alternatively, the difference signal $V_{\mathrm{d,imp}}(n)$ can be added to $V_{\mathrm{cc,filt}}(n)$, for only those values of n at which the peaks of $V_{\mathrm{cc,opt}}(n)$ occur, in order to meet the peak power demands of the output signal and to avoid clipping. But in doing so, the output signal will get much distorted, since the errors introduced by bandwidth reduction at other positions cannot be completely corrected by adjusting the input RF signal.

The best approach is to select those locations or the values of n at which:

$$V_{\mathrm{cc,filt}}(n) < V_{\mathrm{cc,opt}}(n)$$

so that,

$$V_{\mathrm{d}}(n),\ V_{\mathrm{d,imp}}(n) > 0$$

The difference signal $V_{\mathrm{d,imp}}(n)$ should be added to the filtered signal $V_{\mathrm{cc,filt}}(n)$ for only those values of n. This technique is being used in the suggested method.

Block Diagram

The block diagram for dual filtering based method is shown in Fig. 18:

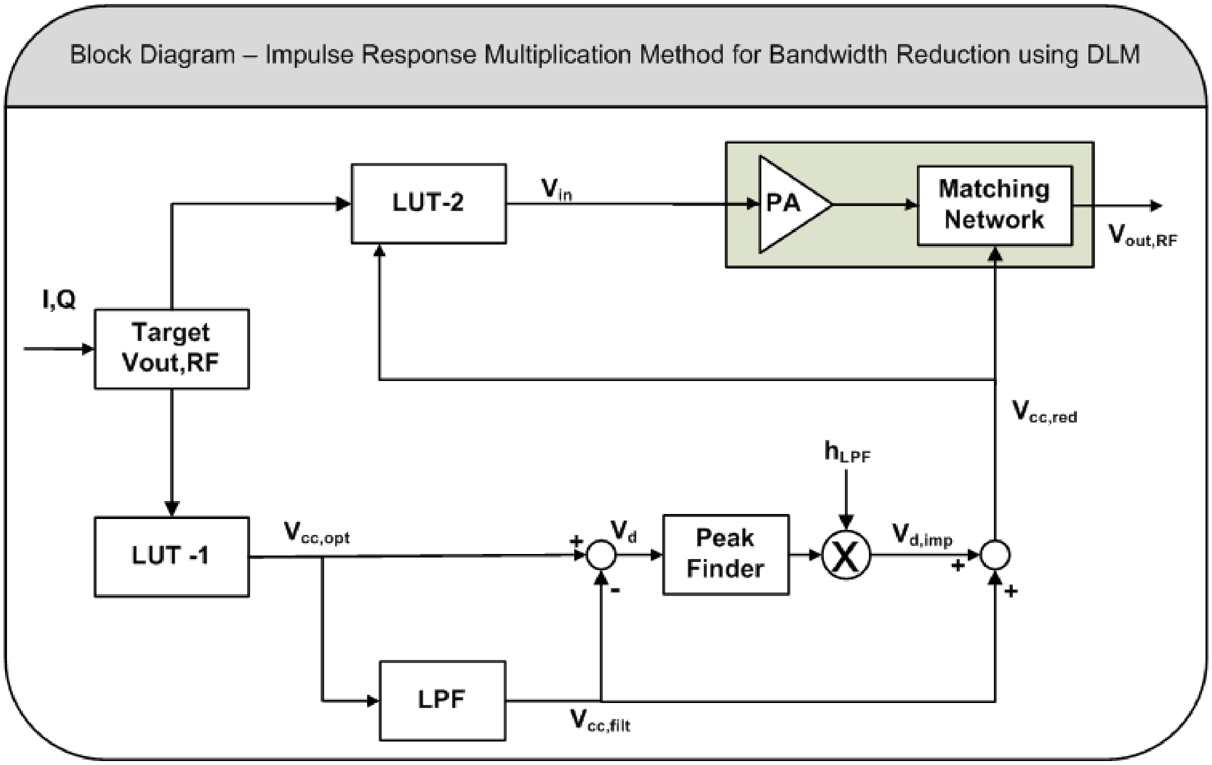

Figure 18: Block diagram for dual filtering method

Description of block diagram

In the block diagram presented in Fig. 18, it can be seen that after low-pass filtering the optimal control signal, the difference of unfiltered and filtered signal is taken as in equation (3.2). The difference signal is then passed through the peak finder block to get the locations containing the peaks or half wave rectifier can also be used to get the positions where filtered signal is lower in amplitude than the unfiltered signal.

After that, a sliding window having size equal to the length of the impulse response of the low-pass filter is applied to the difference signal .The window checks for the central sample and if it is positive valued, it multiplies the sample value with the normalized impulse response of low-pass filter and then it shifts over the next sample. The window passes over the entire signal and all the locations having positive differences are fixed in this manner.

The resulted signal is then added to the filtered signal to get the reduced bandwidth signal $V_{\mathrm{cc,red}}$ as stated in equation (3.4). The plot of reduced bandwidth signal is shown versus the original envelope signal is shown in Fig. 19.

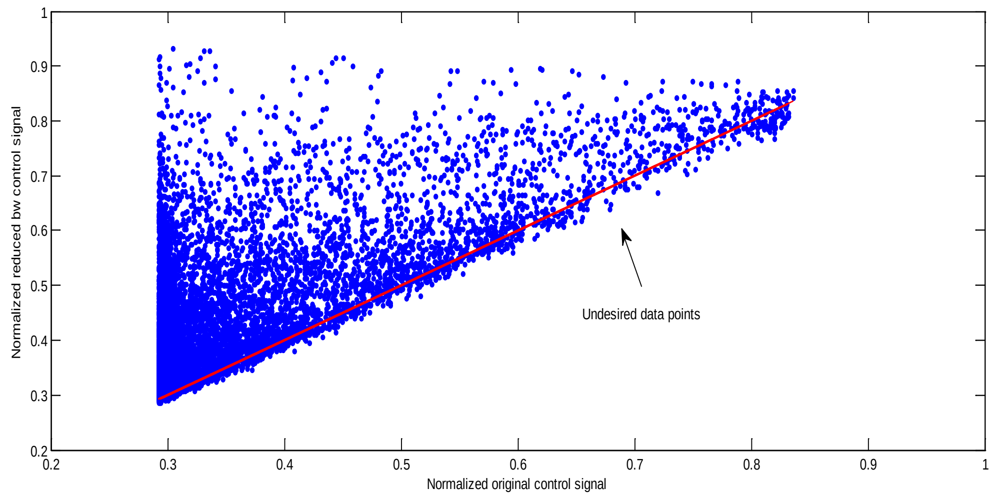

Figure 19: Plot of normalized original signal vs. normalized reduced bandwidth signal before iteration

It can be seen in Fig. 19 that many samples lie below the original envelope and thus violating the equation (3.1) which can cause the distortion at the output of the DLM transmitter. This is due to the fact that when the impulse response is multiplied with the central sample, the adjacent samples also get affected. To fix the other samples, the reduced bandwidth signal is needed to be iterated again and several iterations are required to get rid of all the samples below the original envelope.

The iterations are performed as illustrated in Fig. 20.

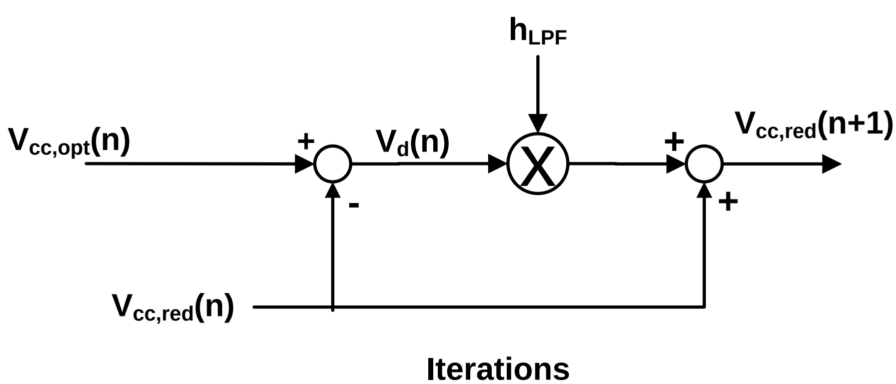

Figure 20: Block diagram showing iterations

The reduced bandwidth signal $V_{\mathrm{cc,red}}(n)$ is applied back to the difference block to get $V_{\mathrm{d}}(n)$, the remaining steps are same as in previous case and finally the iterated signal $V_{\mathrm{cc,red}}(n+1)$is obtained at the output. $'n'$ denotes the number of iterations. In this case, around 1-2 iterations were enough to fix all the samples and it strictly satisfies equation (3.1) as can be shown in Fig. 21:

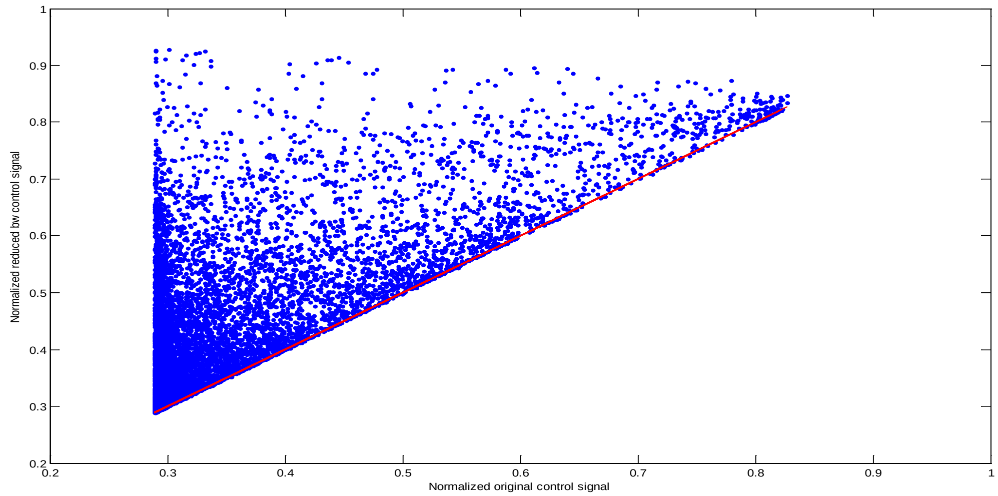

Figure 21: Plot of normalized original signal vs. normalized reduced bandwidth signal after iterations

After getting the reduced bandwidth control signal, it is then applied to the LUT-2 along with the target output signal. LUT-2 is a two-dimensional lookup table and it gives the required RF input signal needed to produce the linear output when applied to PA and matching network block along with the same reduced bandwidth control signal.

Simulation Plots

In simulations, the bandwidth of baseband control signal is reduced to 4 MHz. The signal plots at various stages of the simulated model are shown in Fig. 22:

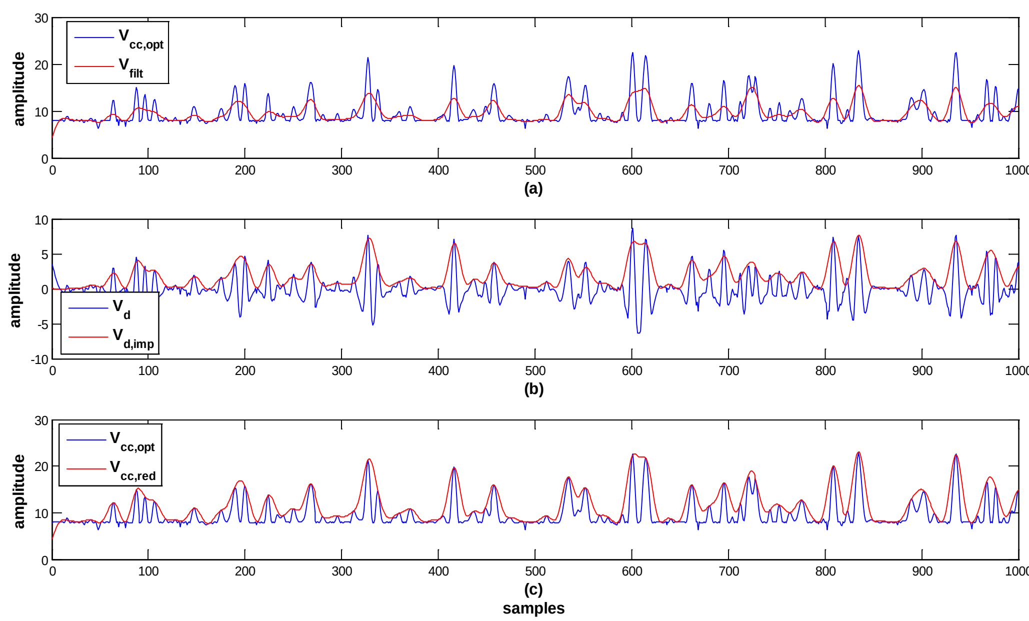

Figure 22: Signal plots at various stages for dual filtering method

- Fig. 22(a) is showing the original envelope of the optimum baseband control signal and its filtered version. It can be seen that although the filtered signal is reduced in bandwidth, but it also has smaller amplitude than the original envelope which causes distortion and clipping in output signal.
- In Fig. 22(b), the difference of the original and the filtered signals in Fig. 22(a) is taken and then this difference signal is multiplied with the impulse response of low-pass filter which is shown in red.
- The resultant signal obtained from Fig. 22(b) is added to the filtered signal shown in Fig. 22(a) to get the final reduced bandwidth signal which is shown in Fig. 22(c) along with the original envelope signal. The reduced bandwidth signal is satisfying the condition stated in equation (3.1).

The PSD plots of original and reduced bandwidth control signals are shown in Fig. 23. The bandwidths are measured at 99.99% energy.

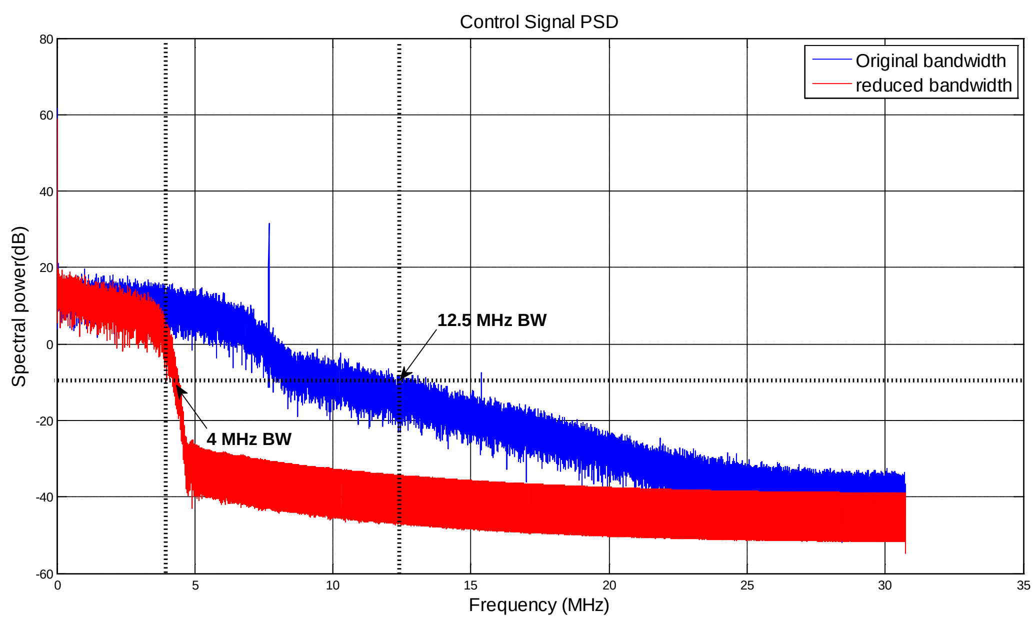

Figure 23: PSD plots for original and reduced bandwidth control signal and target and achieved output signals

The original bandwidth of optimal control signal is 12.5 MHz and it is reduced around 3 times to 4 MHz which can be seen in above plots clearly. The PAE achieved is 50.3% with only 4% reduction compared to the ideal efficiency.

Furthermore, the efficiency achieved at different bandwidths of baseband control signal is compared in Fig. 24.

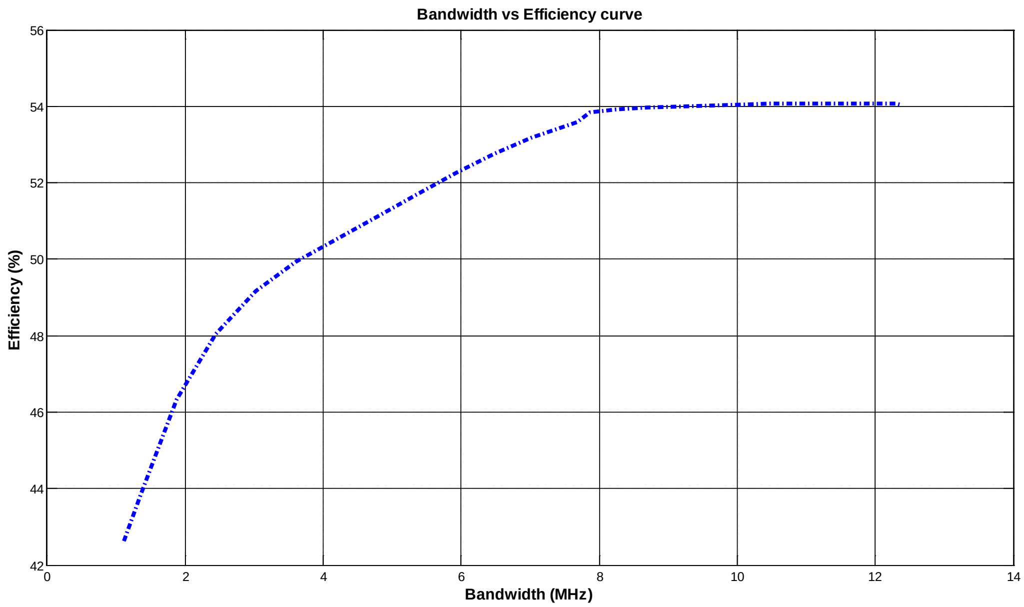

Figure 24: PAE versus Bandwidth curve

It can be seen in Fig. 24, there is not any degradation in efficiency up to 8 MHz and it is almost constant around 54%. Up to 3 MHz, the efficiency reduces gradually and it is around 47% at 3 MHz. Below that bandwidth, efficiency reduces drastically and it is around 43% at 1.5 MHz bandwidth.

Chapter 4 Comparison between different schemes

The methods discussed in this thesis for bandwidth reduction are compared in terms of design complexity, accuracy and efficiency achieved.

Design complexity

By comparing different schemes in terms of design complexity, it is found that maximum value filter method has simpler design. Since the other method requires iterations, so the design is more complex and it needs to calculate the number of iterations to get the accurate bandwidth reduced signal.

However, in maximum value filter method, it is required to calculate the correct order of the maximum value filter to meet the peak power demands of output signal. The order of maximum value filter depends on the cutoff frequency and length of impulse response of the low-pass filter. So to find the correct order, it can initially be set to any value, and then that value is incremented in small steps and it is checked at each step if it satisfies the condition stated in equation (3.1).

Run- time complexity

The run-time complexity of any technique can be measured by the number of operations involved in that technique. Although, the design of maximum value filter method is simpler, however, it takes plenty of time to run. The most of the time is consumed in calculating the order of the max value filter. Since the order of max value filter is incremented in small steps in a loop and at each step, the reduced bandwidth signal is compared with the optimal signal to verify the condition stated in equation (3.1). If the condition is fulfilled the loop terminates, otherwise, it continues to run in this way. So for low bandwidth reductions, the method runs with good speed but it becomes very slow for high bandwidth reductions.

The numbers of operations involved in dual filtering method depends on the technique being used to implement the filter for the rectified signal. The window based multiplication is a little complex, since the multiplication window moves over each sample of the signal; hence it requires more number of operations than using the FIR filter.

Efficiency

The different techniques discussed above for bandwidth reduction of baseband envelope signal can be compared in terms of PAE achieved at different bandwidth reductions in a same plot. The comparison is shown in Fig. 32:

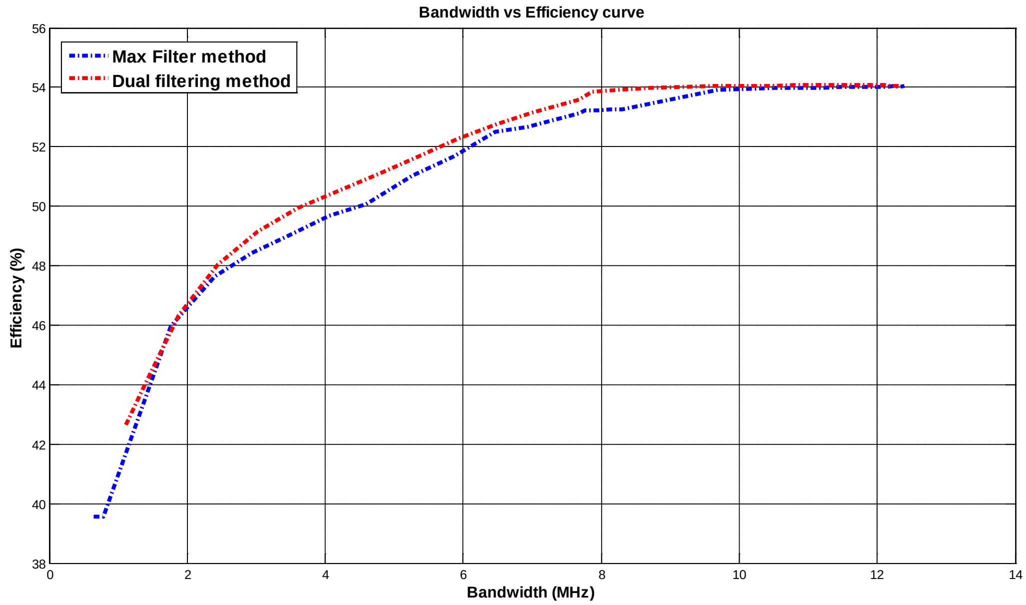

Figure 25: Comparison curve for efficiency achieved at different reduced bandwidths

In Fig. 32, it can be seen that efficiency is almost constant up to 8 MHz without any degradation by using any of the techniques. However, form 2 MHz to 8 MHz, method based on dual filtering has slight high efficiency than the other technique. It is even possible to reduce the bandwidth as low as 1 MHz by using max value filter method, but the efficiency is reduced severely and becomes 39%. So in a big picture, dual filtering method shows high efficiency and more accuracy in delivering the control signal.

The zero bandwidth is also an interesting case to check the performance of the controlling scheme being used. So instead of using the modulated envelope signal, a constant dc level is used with amplitude equal to that of maximum peak of optimal envelope signal, to meet the condition stated in equation (3.1). With such a zero bandwidth signal, we achieved 38% efficiency with which shows 16% degradation in comparison with ideal efficiency.

Chapter 5 Conclusion and Future Work

In this thesis work, different techniques are presented for the bandwidth reduction of baseband control signal and the results are shown in Matlab. The schemes investigated in this project, have the potential to cope with wider bandwidth signals used in 4th generation networks and advanced modulation formats. They have the ability to effectively reduce the bandwidth of baseband envelope signal up to three times, thus simplifying the transmitter design and without reducing much efficiency and causing distortion at the output. These techniques have a wide range of applications and can also be employed in other similar type of transmitter architectures as well, e.g. ET and EER.

These techniques will be tested in real measurements in the future. In real measurements, the issue of memory effect will be encountered which causes distortion and non-linearity at the output. However, it can be fixed by designing proper memory based digital pre-distorters.

The non-linear elements present in PA and matching network cause the spectral re-growth which give rise to distortion components such as third order inter-modulation products(IM-3). If the transmitter system contains energy storage elements such as capacitors and inductors, it will cause memory to the system. In the frequency domain, the consequence of memory is seen as a frequency-dependent gain and phase shift of the signal.

As a result, the IM3 products are not constant but vary with the signal bandwidth and amplitude and cannot be cancelled by simple memory-less linearization techniques. This phenomenon causes distortion and is called memory effect i.e. the current output of the PA depends not only on the current input, but also on past input values. In other words, PA becomes a nonlinear system with memory.

For such a PA, memory-less pre-distortion cannot achieve very good performance. Therefore, digital pre-distorters also need to have memory structure and should be capable of linearizing PAs with memory effects.

References

[1] Frederick H. Raab *"High-Efficiency Linear Amplification by Dynamic Load Modulation"* IEEE MTT-S Digest 2003

[2] Mashad Nemati, ; Fager, Christian; Gustavsson, Ulf; Zirath, Herbert *"An Efficiency Optimized Controlling Scheme for Dynamic Load Modulation of Power Amplifiers"* IEEE Trans. Microwave Theory and Techniques, 58 (4) pp. 873-881.

[3] Mashad Nemati, Hossein; Fager, Christian; Gustavsson, Ulf; Jos, Rik; Zirath, Herbert *"Design of Varactor-Based Tunable Matching Networks for Dynamic Load Modulation of High Power Amplifiers".* Microwave Theory and Techniques, IEEE Transactions on, 57 (5) pp. 1110-1118.

[4]Björn Almgren *"Dynamic Load Modulation"*, Master's Thesis, University of Gavle, Oct 2007

[5]F. H. Raab *"Effects of VSWR upon the class-E RF-power amplifier".* Proc. R F Expo East '88, Philadelphia, PA, pp. 299 - 309, Oct. 1988.

[6] Jinseong Jeong, Donald F. Kimball, Myoungbo Kwak, Paul Draxler and Peter M. Asbeck "*Wideband Envelope tracking Amplifiers with reduced bandwidth power supply waveforms and adaptive digital pre-distortion techniques"* Microwave Theory and Techniques, IEEE Transactions pp. 3307-3314

[7] Albert Cesari, Angel Cid-Pastor, Corinne Alonso and Jean-Marie Dilhac *"A DSP structure authorizing reduced-bandwidth DC/DC Converters for Dynamic Supply of RF Power Amplifiers in Wideband Applications"* , IECON 2006 , pp. 3361-3366, Nov. 2006.

[8] M. K. Kazimierczuk, "*RF Power Amplifiers"*, John Wiley & Sons, New York, NY, 2008, pp. 1-405.

[9] Janne P. Aikio *"Frequency domain model fitting and volterra analysis implemented on top of harmonic balance simulation"* Academic Dissertation, University of Oulu

[10] Lei Ding *"Digital Predistortion of Power Amplifiers for Wireless Applications"* PhD thesis, Georgia Institute of Technology.

[11] Joel Vuolevi, Timo Rahkonen. "*Distortion in Rf Power Amplifiers".* Artech House Publishers , 2003.