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ANALOG CHAIN CALIBRATION IN DIGITAL BEAM-FORMING APPLICATIONS

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ABSTRACT

This paper deals with the issues present in a Digital Beam Forming Network (DBFN) system related to the non-idealities introduced by the hardware implementation. In particular, the paper considers the mismatches of the different electronic chains (corresponding to the different beams) used for the signal conditioning and acquisition. In this work a new algorithm for the chain calibration is proposed and analyzed. This algorithm has been developed for obtaining three fundamental advantages with respect to other methods proposed in the literature. The algorithm doesn't take any assumptions regarding the primary signal structure; it can be applied during the normal working of the system and doesn't require any initial calibrations. The algorithm is based on the injection of a low level pseudo-noise signal that is recovered and used for the training of an adaptive filter.

Keywords: DBFN, DOA, calibration, antenna array, digital compensation.

INTRODUCTION

In recent years, the "smart antennas" have gained a lot of interest for their ability to improve the performance of radio systems. The principles the smart antennas exploit are not new for the world of telecommunications, but only recently, due to technology progress, they are diffusing in the commercial systems. Smart antenna is a system composed of a homogeneous set of antennas arranged in space and combined with a suitable processing of the signals. More information on the use of array of antennas in telecommunication systems can be found in (Chryssomallis M. 2000). The two essential tasks in a smart antenna implementation are the estimation of the Direction of Arrival (DOA) of the incident signals and the Beamforming for the transmitting and receiving signals. These tasks are fundamental both for the enhancement of the useful signals and for the suppression of the disturbing interferences. In the actual systems the presence of undesired phenomena (in particular the errors related to the implementation) can degrade the performance until the failure condition. Possible sources of errors in actual systems are the following:

a) Positioning errors.

b) Electromagnetic coupling.

c) Mismatch errors in the sensors.

d) Errors in the processing chain.

The first problem is connected with the real position in space of the antenna sensors. For example, in a linear array the elements should be placed on straight lines at regular intervals. However, for the inevitable errors related to mechanical tolerances of the manufacturing process, the positions of these elements are different with respect to the nominal positions. Using some calibration algorithms (Boon Chong Ng *et al* 1996), (Hung, E. K. L. 2000) it is possible to determine the actual location of the sensors and then calculate the steering vector. In a calibration process these positioning errors can be modelled as a phase delay that can be associated to the signal transduction chain.

The second error is related to the antenna's interactions. Since the elements are positioned at distances similar to the signal wavelengths, strong phenomena of electromagnetic coupling are present. Many authors consider this phenomena, together with the gain and phase errors introduced by the chains, the most harmful errors and they proposed appropriate calibration techniques to solve this issue (Aumann, H. M. and Willwerth, F. G. 1995), (Dandekar, K. R. & all 2002), (Yan Wang & Shanjia Xu 2003). The mismatch between the sensors produces phase and amplitude errors on the sensor outputs.

The last source of error is related to the transduction and processing chain which is composed of different components (amplifiers, down converters and samplers with A/D converters) connected by metal lines. The simplified model, normally used for the analogue chains (that treats the non idealities as attenuation and phase shifting), considers, as fundamental hypothesis, the narrow band assumption. When broadband systems are analyzed, this model introduces errors. In fact, some elements of the chains (for example sampling and hold circuits) have frequency response not constant in the band of interest.

The above analysis allows concluding that in manufactured arrays there are parameters that are different with respect to the nominal ones. Since most DOA and Beam forming algorithms in the literature are based on the assumption that the input data come from a perfectly characterized and ideal array, the use of these algorithms on real systems requires the calibration of the structure in order to obtain a high-precision electronic pointing of the antenna (Harter, M *et al* 2015).

In this work, we analyze the characteristics of the acquisition chains, defining the main parameters that potentially could impact on the performance of the beam forming network. Starting from this preliminary analysis the authors have developed a model for the processing chain. This model is then used for developing a new calibration methodology. In the following sections the chain model, the calibration method and the simulation results are presented and discussed.

DIGITAL BEAMFORMING BASICS

In our experiments we consider a linear array of M antennas at a distance d. In Figure-1 a plane wave with angle φ hits the antenna array. The signals between two elements are delayed of t = d*sin(φ)/c, where c is the speed of light.

If we combine synergically the M signals coming from the M antennas (see Figure-2) delaying and adding them in a single output signal, we can produce additive interferences in some directions and destructive interferences in other directions.

This is the principle of the Beamforming technique. If we can suppose narrowband signals, the delays shown in the scheme of Figure-2 can be replaced with simpler phase shifters (implemented with complex multipliers).

In such a scheme we can define three elementary blocks:

a) The antenna array.

b) The beamforming network (analog or digital).

c) The processor (used for the computation of the DOA and the beamforming coefficients).

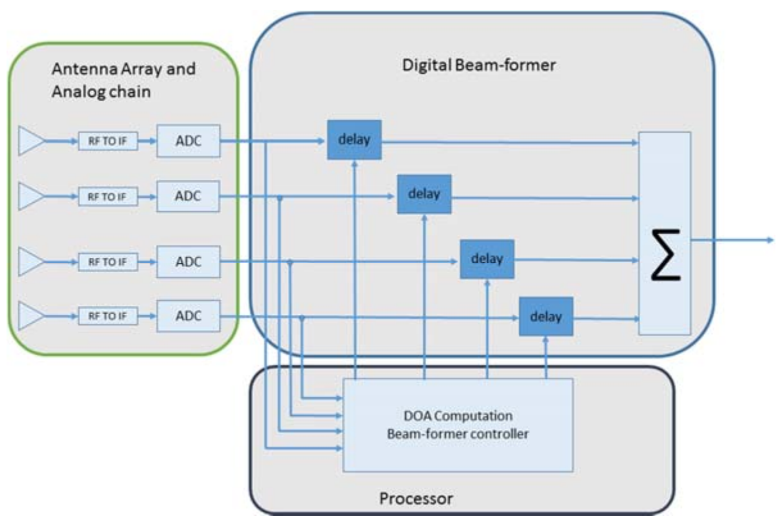

Figure-1. Beamforming system.

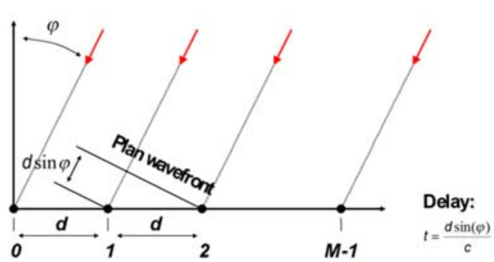

Figure-2. Interaction of a plan wave with an antenna array.

The beamformer scheme of Figure-2 is generic, suitable for representing either analog or digital beam-forming systems. The digital solution presents some important advantages, as the possibility to implement variable and accurate delay blocks. These blocks are mandatory if wide-band signals must be processed and they can be implemented using fractional delay filters (Laakso, T. I., *et al* 1996). The calibration procedure proposed in this paper is valid either for narrow-band and wide-band systems. As previously introduced, the different sections of the actual systems are affected by different non-idealities (Litva, J. and Yeung Lo, T.K. 1996). In this section we analyze the effects of these non-idealities on the frequency response of the chain (phase and amplitude distortions).

Non idealities of antenna array

An ideal antenna array is characterized by a perfectly known spatial distribution of elements on an axis (in the linear case) or on a plane (in the case of planar or circular arrays).

However, the actual structure of an array can be characterized by physical inaccuracies and electrical properties of different nature among which we can mention:

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- Positioning errors of the array elements.
- Rotation of the elements around the axis normal to the plane of antenna element.
- Non rigidity of the structure sustaining the array.
- Mutual coupling between the antenna elements.
- Length differences between the various channels transporting the signals (wave guides, metaltracks).

These non-idealities introduce frequency dependent errors on the amplitude of the acquired signal.

Non idealities of the analog section

Even in the devices of the transceiver section are present non-idealities that introduce phase and amplitude errors on the signals. The non-ideal components, that can limit the system performance, are summarized in the following list:

- RF band-pass filter for the suppression of the frequency image.
- Low-noise amplifier (LNA).
- Down-conversion stages, implemented with mixers.
- Low-pass filters at the output of the mixers.
- Additional amplification stages.

All the active devices that compose the receiver section are characterized by the non-linearities or distortions listed below:

- Gain compression.
- Intermodulation products (first and third order).
- Self-noise.
- Generation of harmonic spurious at the output of mixers.

Non idealities of digital processing section

The digital processing section is basically composed of:

- An Analog to Digital Converter (ADC).
- A processor for the Digital Signal Processing (DSP).

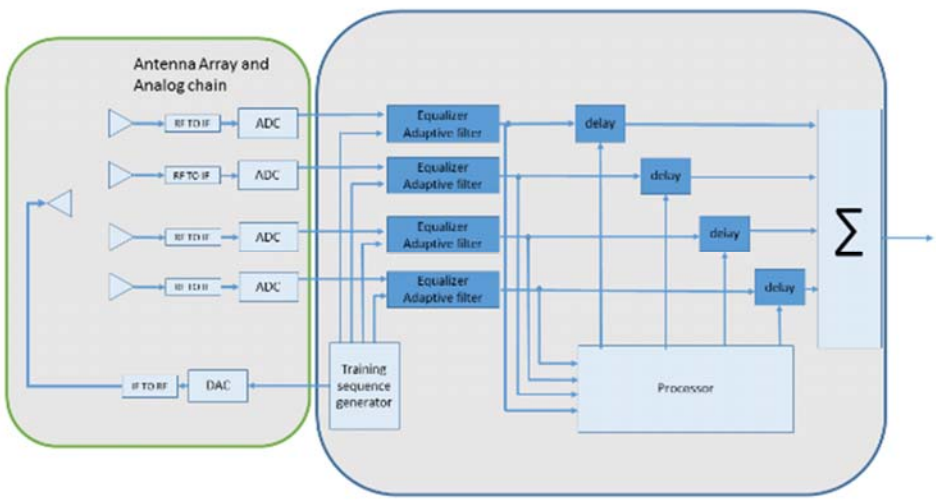

Figure-3. Structure of DBFN with equalization system.

On the basis of the devices used in this section, the introduced errors are the following:

- Quantization noise.
- Saturation effects.
- Skew and jitter of the control signals of the ADC.

Proposed calibration method and simulation mode

In order to improve the performance of the system (in terms of directivity and interference suppression) we need to reduce the effects of the non-idealities (Harter, M *et al* 2015)

A possible method for compensating such effects is the calibration of the array. There are different calibration methods proposed in literature. Some methods consider specific applications. For example, (Harter, M *et al* 2015) discusses a calibration procedure specific for radar applications. Moreover, in the past some works have been published for narrow-band calibration, see for example (Hampson, G. A., Smolders, A. B. 1999) and (Nuteson, T. W *et al* 2006). At the present, the developing of calibration methods for wideband digital beamformers is an emerging research field (Adithya *et al* 2014), (Bruckmeyer J. P., Kostanic I. 2015). For example in (Bruckmeyer J. P., Kostanic I. 2015) an adaptive filter is used for the calibration. All the above solutions cannot perform calibration during the normal operation of the system and, in some cases; they have been developed for specific input signals.

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The objective of this paper is to propose a method able to perform an on-line calibration (without any service interruption) for generic input signals.

Our method performs this calibration using a sequence superimposed on the user signal during the normal work of the system. This sequence is then extracted and used for the training of an adaptive filter, responsible for the compensation of the above non-idealities.

The calibration/equalization process is carried out independently on each individual chain. The resulting structure is shown in Figure-3. The figure shows the training sequence generator which sends the data either to the adaptive filter or to a calibration horn, which injects the training sequence in the different chains.

The equalization algorithm is based on the definition of a reference FIR filter representing the expected frequency behavior of the single chain. The main task of the procedure is to insert an adaptive filter in the chain, such that the overall frequency response of the chain will approximate, as close as possible, the response of the reference filter.

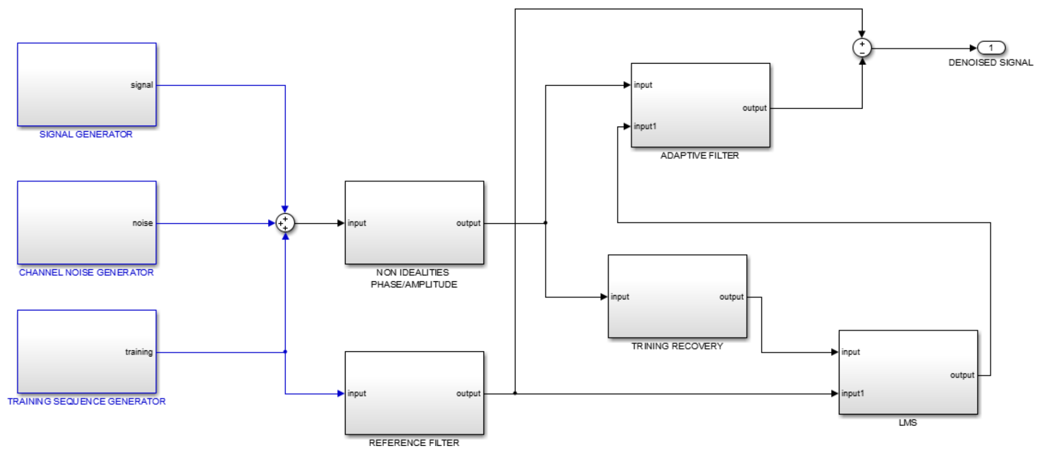

Figure-4. Simulink implementation of DBFN system.

Figure-4 shows the structure of the simulation model representing the proposed system. A suitable pseudorandom training sequence is generated by the "Training Sequence Generator" and is injected in each antenna. The level of this sequence is very low with respect to the level of the incoming signal (generated in the model by "Signal Generator" block). The low level assumption makes the sequence effects negligible for the communication system and, consequently, the calibration procedure can be performed during the normal working.

Due to the non-idealities, the signal received at the output of analog chain contains a distorted version of the sequence. This sequence is extracted from the signal (in the "Training Recovery" block) and is used for the training of the adaptive filter, whose coefficients are computed by the block implementing the Least Mean Square (LMS) algorithm.

Different simulations have been performed using the Simulink model of Figure-4. In this model the input signal is obtained adding the two above discussed components, the pseudo-noise training sequence signal and the incoming signal.

Since the calibration must be independent of the type of signal present in the incoming signal (i.e., spectrum shape, modulation type, etc), in our simulations we modeled the input signal as random noise.

The non-idealities of the analog chain have been emulated by digital filter (for the effects in frequency) and a variable fractional delay (for the delay mismatches).

The output signal of the "Non-Idealities" block is then sent to the receiver section where the conventional traffic signal is extracted and equalized. The receiver is composed of two branches. The lower branch in Figure-4 generates the reference signal (filtering the recovery sequence with the reference filter), it extracts the training signal (in the "Training Recovery" block through the computation of the average) and it compares the reference signal with the extracted signal for computing the new equalization filter coefficients (these coefficients are computed by the "LMS Block"). The "Adaptive Filter", present in the upper branch, equalizes the received signal. The training sequence is finally eliminated by subtracting the sequence at the output of the "Reference Filter" from the equalized signal (the output of "Adaptive Filter"). This signal is the output of the equalization block and it's used by the beamforming processor.

The performance of the method described above was evaluated through a set of simulations. A main objective of these simulations was the evaluation of the quality of the final output signal. An important aspect is to estimate how the power of the pseudo-noise sequence impacts on the calibration results. In fact, the reduction of the pseudo-noise sequence amplitude allows obtaining less interference on the main signal but limits the calibration precision.

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ALGORITHM DESCRIPTION

The two main elements of the equalization algorithm are the training sequence recover and the LMS computation blocks. These two elements are described in the following.

Training recovery

In the proposed approach, based on a continuous on-line calibration, the training sequence has very low amplitude in order to reduce the interference with the main signal. For this reason the ratio between the power of the training sequence and that of the main signal is very low (in the paper we call this ratio SNR_{IN}). As a consequence the first section in the receiver is the "Training Recovery" block that extracts the training sequence from the complete signal through an averaging process. It extracts the training sequence modified by the effects of the above discussed non-idealities present in the channel (for a single chain).

In the following algorithm we suppose the synchronization of the data acquisition with the starting of the pseudo-noise sequence. It's important to underline that the synchronization, due to the periodicity of the pseudo-noise sequence, can be also performed after the training recovery. The input data are divided into vectors of length P (where P is the length of the training sequence). The average is performed on the rows of the matrix. This method is described in Equation. 1

$$\begin{bmatrix} x_0 & x_P & x_{2P} & \cdots & x_{RP} \\ x_1 & x_{P+1} & x_{2P+1} & \cdots & x_{RP+1} \\ \vdots & \vdots & \vdots & \ddots & \vdots \\ x_{P-1} & x_{2P-1} & x_{3P-1} & \cdots & x_{(R+1)P-1} \end{bmatrix} \longrightarrow \hat{d} =$$

$$\frac{1}{R}\begin{bmatrix} \sum_{i=0}^{R} x_{iP} \\ \sum_{i=0}^{R} x_{iP+1} \\ \vdots \\ \sum_{i=0}^{R} x_{iP(i+1)-1} \end{bmatrix} = \begin{bmatrix} d_0 + E[u(nP)] \\ d_1 + E[u(nP+1)] \\ \vdots \\ d_P + E[u(P(n+1)-1)] \end{bmatrix} \quad (1)$$

where E[.] represents the expected value operator. The performance of the training-recovery block is evaluated though the signal to noise ratio at the output, SNR_{out}, defined as:

$$SNR_{out} = Pi/Pe \quad (2)$$

where Pi is the power of the received training sequence modified by the non-idealities of the channel and Pe is the power of the difference between the training sequence mentioned above and that recovered through the averaging process. Pe represents the power of the residue of the main signal not cancelled by the averaging operation. Increasing the number of averages, improve the quality of the reconstructed sequence. Figure-5 shows the number of iterations required for obtaining a given improvement of the SNR, $\Delta SNR = SNR_{out} - SNR_{in}$. The number of iterations grows exponentially with the increasing of ΔSNR, independently of the SNR_{in}. The recovered training sequence is then sent to the following sections of the equalizer.

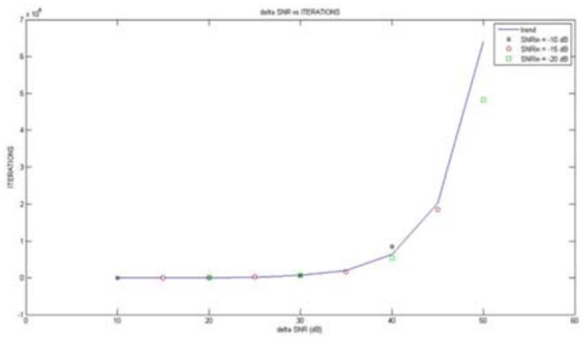

Figure-5. Number of iterations vs. Δ SNR.

LMS algorithm

The adaptive filter can be implemented in different ways and with different hardware architectures (Bernocchi *et al* 2007). In our implementation the training of adaptive filter is based on LMS algorithm. The LMS algorithm is realized in three phases:

a) Computation of the output signal y(n) produced by the filter using the coefficients computed in the previous adaptive step

b) Computation of the error estimate e(n), comparing the filter output with the reference signal d(n).

c) On the basis of the error computed in the point 2, the adaptive system calculates the new coefficients of the filter in order to minimize the error.

Analytically the above three phases can be described as follows (where bold characters identify vectors):

Filtering: $y(n)=\mathbf{w}^T(n)\,\mathbf{x}(n)$

Error estimates: $e(n)=y(n)-d(n)$

Adaptation of the coefficients: $\mathbf{w}(n+1)=\mathbf{w}(n)-2\mu\, e(n)\,\mathbf{x}(n)$

Updating of the coefficients: $\mathbf{w}(n)=\mathbf{w}(n+1)$

The result of the adaptive procedure depends on the value of the constant μ.A high value of μ allows a rapid convergence of the algorithm but with a greater error. Viceversa, if μ is very small, the convergence will require a greater number of iterations but the steady-state error will be reduced.

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SIMULATION RESULTS

As described above, the equalization procedure starts with the definition of a reference filter. This filter represents the desired frequency-behaviour of the full chain. As a consequence, after the equalization, the chain (composed of antenna, analogue sections and equalization filter) must have a frequency response as close as possible to the reference filter. The frequency-response (magnitude and phase) of the reference filter used in our experiments is shown in Figure-6 (green line). In the same figure the response of the actual channel (before equalization) chosen for our simulations is also represented (blue line).

The main parameters of the above reference filter and the unequalized channel of Figure-6 are summarized in Table-1 and Table-2, respectively.

Table-1. Parameters of the reference flter.

Maximum value in passband	**0.501 dB**
Minimum value in passband	-0.319 dB
Maximum variation amplitude in passband	0.821 dB
Minimum attenuation in stop band	-26.67 dB

Table-2. Parameters of the channel before equalization.

Maximum value in passband	**2.30 dB**
Minimum value in passband	-1.83 dB
Maximum variation amplitude passband	4.13 dB
Minimum attenuation in stopband	-5.93 dB
Spread phase in the passband (respect to the reference)	33 degrees

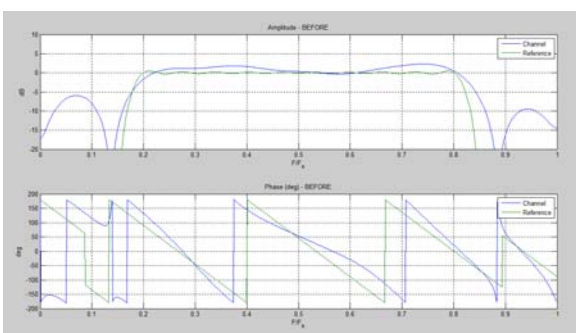

Figure-6. Reference flter (in green) and channel frequency response before equalization (in blue).

Different simulations were carried-out for evaluating the performance of the blocks composing the proposed system. The first block to evaluate is the training recovery block. For this block the SNR_{out} can be improved increasing the length of the average (see Figure-5). In our experiments we performed different simulations using a different number of averages such that SNR_{out} assumed four different values: 0 dB, 10 dB, 20 dB and 30 dB.

In the following, the equalization results for SNR_{in} = -10dB and for the above different values of SNR_{out} are described. For each SNR_{out} the equalization results are summarized in Table-3 the frequency responses (amplitude and phase) are also plotted in Figure-7 and Figure-14. The plots show the reference response, the response before equalization and response after equalization, for each value of SNR_{out}.

In Figure-15, Figure-16, Figure-17 the calibration results for different values of SNR_{in} are summarized using a set of plots. In these plots the value of the ripple amplitude, phase difference and number of iterations are presented for different values of SNR_{in} (-10 dB, -15 dB and -20 dB) and SNR_{out} (0 dB, 10 dB, 20 dB, 30 dB).

CONCLUSIONS

In this paper a method for the equalization of the processing chains in an antenna array of a digital beam forming system is presented. This equalization is necessary for improving the characteristics of the formed beam. The technique is based on pseudo-noise sequences injected at very low-level inside the user signal. In this way equalization can be performed without any service interruptions. Pseudo-noise sequences are then recovered through averaging. The number of averages depends on the ratio between the power of the training sequence and that of the user signal. The recovered signal is used for the training of the adaptive filter Simulations were performed for evaluating the effect of pseudo-noise sequence level on the equalization performance. The simulation results confirmed the validity of the proposed technique and its adaptation to different operative scenarios. In particular increasing the number of iterations, it is possible to maintain the ripple amplitude and phase errors below 1.7 dB and 3 degrees.

Due the processing structure of proposed algorithm, based on conventional blocks as averagers and adaptive filters, it is very suitable for hardware implementation. In particular solutions based on reconfigurable and programmable devices (as for example hybrid architecture based on FPGA and processor) (Cardarilli GC *et al* 2008), (Cardarilli GC *et al* 2009), (Cardarilli GC *et al* 2010) can be used for implementing the algorithm in very efficient way.

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Table-3. Equalization results for SNRin = -10dB.

SNRout	0 dB	10 dB	20 dB	30dB
Processed frame number	459	6,602	52,723	845,565
Maximum value in passband	-4.75 dB	4.68 dB	2.66 dB	0.64 dB
Minimum value in passband	-31.20 dB	-2.80 dB	-1.62 dB	-0.56 dB
Maximum variation amplitude in passband	26.44 dB	7.49 dB	4.29 dB	1.2 dB
Maximum amplitude difference with ideal response in passband	31.3 dB	4.74 dB	2.53 dB	0.66 dB
Minimum attenuation in stopband	-22.61 dB	-23.07 dB	-25.52 dB	-25.93 dB
Maximum phase difference with ideal response in passband	8 degrees	5 degrees	5 degrees	3 degrees
Frequency response	Figure.7, Figure.8	Figure.9, Figure.10	Figure.11, Figure.12	Figure.13, Figure.14

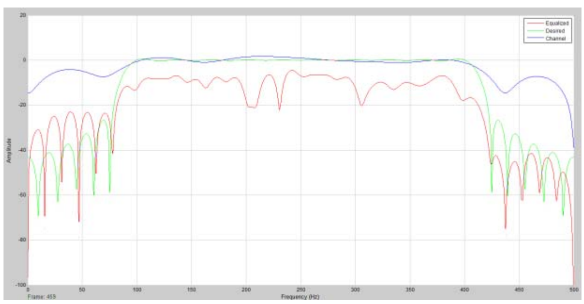

Figure-7. SNRout = 0dB: amplitude response.

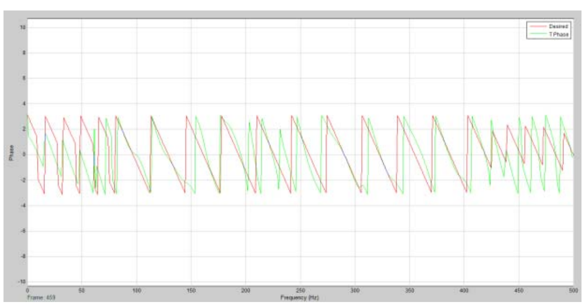

Figure-8. SNRout = 0dB: phase response.

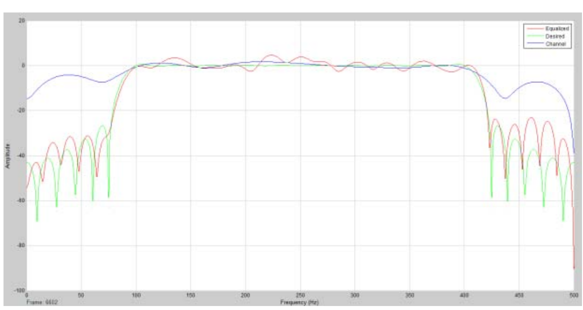

Figure-9. SNRout = 10dB: amplitude response.

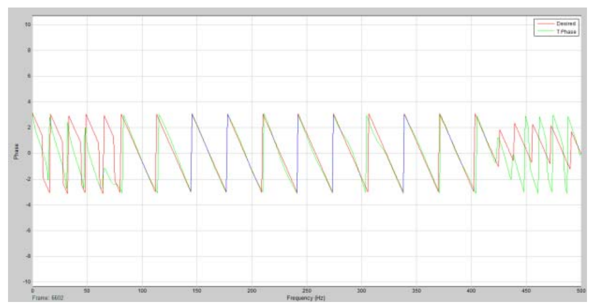

Figure 10. SNRout = 10dB: phase response.

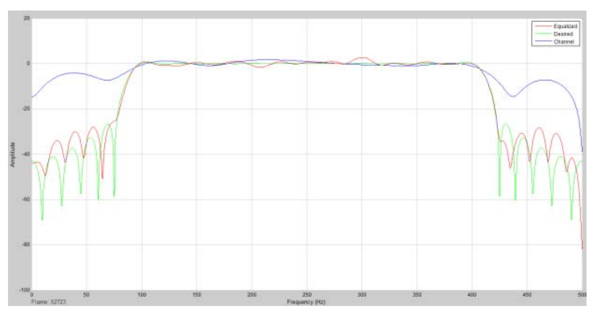

Figure-11. SNRout = 20dB: amplitude response.

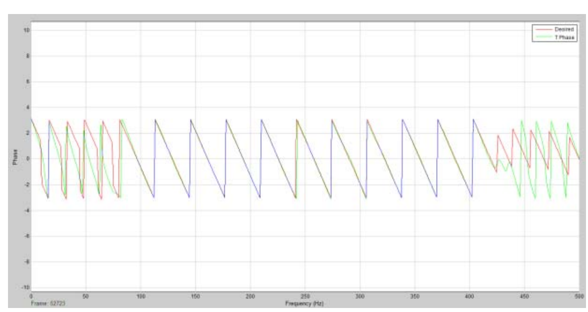

Figure-12. SNRout = 20dB: phase response.

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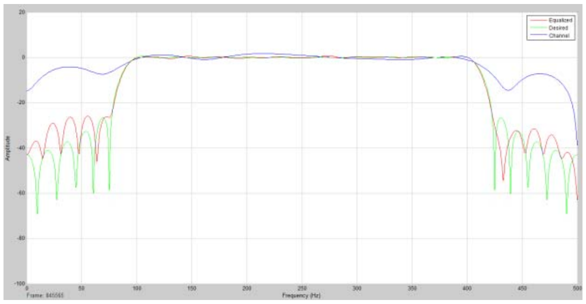


Figure-13. SNRout = 30dB: amplitude response.

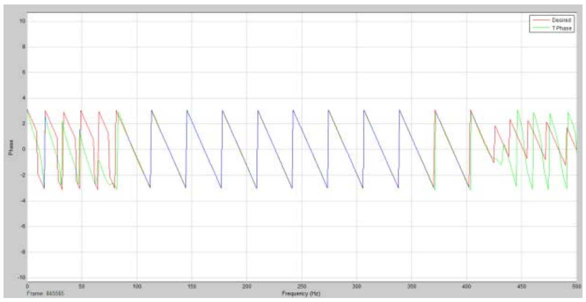
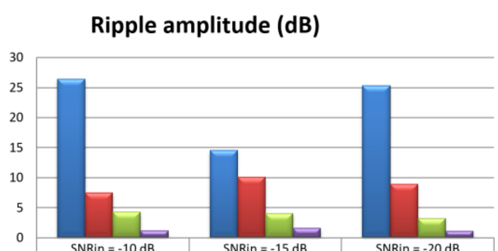
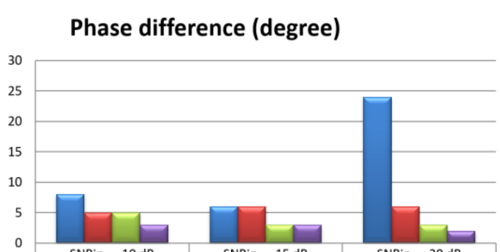


Figure-14. SNRout = 30dB: phase response.



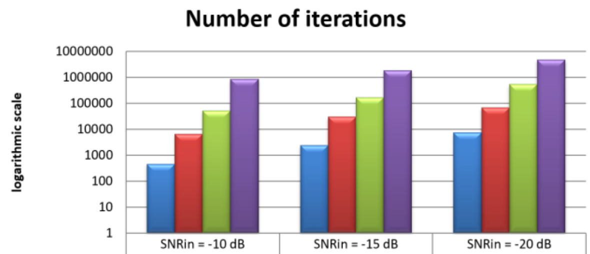
	SNRin = -10 dB	SNRin = -15 dB	SNRin = -20 dB
SNRout = 10 dB	26,44	14,58	25,38
SNRout = 20 dB	7,49	10,09	8,91
SNRout = 20 dB	4,29	4,07	3,24
SNRout = 30 dB	1,2	1,62	1,15

Figure-15. Equalization results: pass band ripple amplitude for different SNR_{IN} and SNR_{OUT}.



	SNRin = -10 dB	SNRin = -15 dB	SNRin = -20 dB
SNRout = 10 dB	8	6	24
SNRout = 20 dB	5	6	6
SNRout = 20 dB	5	3	3
SNRout = 30 dB	3	3	2

Figure-16. Equalization results: maximum phase error for different SNR_{IN} and SNR_{OUT}.



	SNRin = -10 dB	SNRin = -15 dB	SNRin = -20 dB
SNRout = 10 dB	459	2433	7367
SNRout = 20 dB	6602	30634	69237
SNRout = 20 dB	52723	171073	537913
SNRout = 30 dB	845565	1854753	4819827

Figure-17. Equalization results: number of iterations for different SNR_{IN} and SNR_{OUT}.

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