Comparisons of beamforming techniques for 4G wireless communications systems

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Abstract—This document is based on a study initiated in order to evaluate the influence of different beamforming issues on smart antennas functional performances, by comparing several beamforming methods. (Abstract)

Beamforming; smart antennas; wireless; algorithms; (key words)

I. INTRODUCTION

In wireless communication, beamforming is a highly used technique that improves antenna gain and eliminates the interferences. In these circumstances, the needed signals transmit power is reduced. In traditional beamforming methods analog devices are used and they require a lot of customized electronics hardware. Thus, the beamforming implementation costs have been high-priced. In case of digital beamforming, the signal processing is performed through software which significantly increases the flexibility. Moreover, since all electronics are commercial off-the-shelf, expenses of digital beamforming utilities are insignificant comparing to convention.

Although the development of the digital beamforming has been a point of interest for the last 50 years, today we discover its utility in being the solution in many contemporary and forward looking challenges.

Wireless technology is a very dynamic research and innovation field. The 4th generation mobile communication (4G) systems are still looking for feasible approaches for a better power control, suitable spectrum access, intra and inter-base station interference cancellation. Also, the satellite communication technology is in progress and with this incredible increased number of users needs better solutions for its goals.

In the military domain, having digital beamforming as one of its features, the new radar systems placed on satellites answer some complaints about military ballistic missile defense systems, or make new spy and environmental monitoring missions possible from orbit [1].

The sonar systems are known for using digital beamformers, but there are still some needs in quantizing time delays, detecting and modeling the target echo in real-ocean environments [2].

As one can see, the growing interest in an intense research in digital beamforming is fully justified.

II. SMART ANTENNA CONCEPT

A smart antenna is a unit of a wireless communication system. The smart antenna concept is defined as an antenna array with a digital signal-processing capability to operate in an adaptive approach. Also, smart antennas are known as adaptive array antennas or multiple-input multiple-output (MIMO) systems. Their main ability is to determine the direction of arrival (DOA) of the signal, and further on to compute the weights for the digital beamforming process. A smart antenna can automatically modify its radiation pattern as a feedback to the electromagnetic environment. This can improve the performance properties of the wireless system that the smart antenna serves.

Depending on the beamforming strategy, there are switched beam smart antennas and adaptive array smart antennas. Adaptive array is the most advanced smart antenna technology. With proper signal-processing algorithms the antenna is capable to dynamically maximize the desired signal reception by steering the beam to any direction of interest and also by annihilating the undesired signals.

III. BEAMFORMING

Digital beamforming uses different methods according to multiple criteria. Important criterions are in this case the type of applications where beamforming is required, the signal processing speed needed, the angle of arrival of the incoming signals, the complexity of the algorithm. Also, considering these criterions, there are multiple theoretical classifications.

Reflecting about the angle of arrival of the incoming signals, the techniques used for digital beamforming can be group into fixed beamforming approaches and adaptive beamforming algorithms [3]. If the arrival angles don’t change with time, the optimum array weights won’t need to be adjusted. In this case, for fixed arrival angle emitters, the fixed beamforming approaches are to be applied. If the desired arrival angles change with time, it is necessary to design an optimization scheme that operates continuously to keep recalculating the optimum array weights. The receiver signal processing algorithm then must allow for the continuous
adaptation to a permanently-changing electromagnetic environment. The fixed beamforming process is out-of-date and the adaptive algorithm is embraced for its ability to calculate the continuously updated weights. The adaptation process must satisfy a specified optimization criterion.

A few examples of popular optimization techniques include the criterion based upon maximizing the signal-interference ratio (SIR), as fixed beamforming approach. Also, as adaptive beamforming techniques, deserve to be mentioned the least mean squares algorithm (LMS) and the recursive least squares algorithm (RLS).

IV. COMPARISONS OF DIFFERENT BEAMFORMING TECHNIQUES – SIMULATION RESULTS

In order to compare these three techniques, we consider three 9-element arrays, \( d = \lambda/2 \), each one being equipped with one of these algorithms. The sidelobe cancellation capability, the direction selectivity, the influences of placing a null close to a desired signal direction of arrival, the precision, or the complexity of these methods are the characteristics that are to be observed.

It is intuitive that if we can cancel all interference by placing nulls at their angles of arrival, we will automatically maximize the SIR. The signal-interference ratio (SIR) is defined as the ratio of the desired signal power divided by the undesired signal power. The SIR can be maximized by separating the output signal in desired and undesired signals, and nulling the undesired signal. The optimum weight vector and the maximum signal-interference ratio (SIR) can be found by correctly identifying the array covariance matrices for both the desired signal and the undesired signals.

The least mean squares (LMS) algorithm is a gradient based approach. Gradient based algorithms assume an established quadratic performance surface. The solution for the optimum set of weights is predicated on our knowledge of all signal statistics and thus in our calculation of the correlation matrix. In general, we do not know the signal statistics and that is why we must estimate the array correlation matrix and the signal correlation vector. We can employ an iterative technique called the method of steepest descent to approximate the gradient of the cost function and to estimate the array correlation matrix and the signal correlation vector. If we substitute the instantaneous correlation approximations, we have the LMS solution. As the iterative technique depends on the step-size parameter, we just have another coefficient dependency and a greater complexity for this algorithm.

On the other hand the signal sources can vary by changing or moving slowly with time and we might want to erase the earliest data samples and use just the most recent ones.

The future values for the array correlation estimate and the vector correlation estimate can be obtained using previous values. The recursive least squares (RLS) solution is the recursion relationship that can be derived to update the weight vectors.

By using a little data the algorithm is faster and has low complexity, but the forgetting factor introduced must be wisely chosen, or the result won’t be very precise.

For the sidelobe cancellation capability comparison, let us consider for each array one desired signal and two interferers just “squeezing” the incoming desired signal between them. We shall consider a scenario with 2 interferers in symmetric directions relatively to the signal of interest. Although it is a very particular case, this situation is the less convenient one, any other situation being treated in the next scenario, with only one interferer and one desired signal, in order to study the influences of placing a null close to a desired signal direction of arrival.

Although the result is tight, it looks like the LMS algorithm is better than the other two. Considering the desired signal \( \text{AOAd} = 0 \text{ deg} \), the sidelobes increase when the directions of arrival of the interferers are approaching the 0 deg value.

Having the interferers at \( \pm 8.88 \text{ deg} \) (fig. 1), makes the sidelobes level to be 3 dB less than the major lobe level.

The sidelobes level become equal to the major lobe level when interferers are at \( \pm 7.5 \text{ deg} \) (fig. 2).
The sidelobes level is 3 dB greater than the major lobe level for interferers being at ±6.2 deg (fig. 3).

Another parameter observed is the selectivity. Measuring the major lobe width at the 0.707 value from its maximum concludes in very tight results. The measured value is 11.5 deg for SIR and LMS, and also for the RLS, obtained as an average value (fig. 4).

Studying the influences of placing a null close to a desired signal direction of arrival also reveals the LMS algorithm to be better, although the differences spotted could be taking into account depending on the precision wanted for a requested application.

Considering the desired signal AOAd = 30 deg, a null has been placed close to the desired signal direction of arrival. For the major beam to receive only 90% of the desired signal, all the techniques were equal in performances and the null has been placed first time (fig. 5) at AOAi = 22 deg (8 deg difference) and second time (fig. 6) at AOAi = 39 deg (9 deg difference).

For the major beam to become useless and to receive the desired signal reduced with 3 dB, the LMS algorithm was again better, by tolerating nulls closer than the other two methods.

First time (fig. 7), the null has been placed at AOAi = 25 deg (5 deg difference) and second time (fig. 8) at AOAi = 35.5 deg (5.5 deg difference).

CONCLUSIONS

With these measurements and results in mind, the LMS algorithm reveals itself as being more competent than the SIR technique and the RLS algorithm. Still, the other methods had very good results, and deserve credit. It worth mentioned that the MATLAB codes of the LMS and RLS algorithms use the white Gaussian noise (WGN) in their estimations, so the results are never exactly the same for a set of values. Their performances can be appreciated considering the average measured values.

The SIR technique is the simplest method and presents a high efficiency.
The advantage of the RLS algorithm is that it is no longer necessary to invert a large correlation matrix. The recursive equations allow for easy updates of the inverse of the correlation matrix. The RLS algorithm also converges much more quickly than the LMS algorithm.

Considering the requests of the 4G technology, to establish a suitable link between stations and its users characterized by a very high mobility, adaptive algorithms are more appropriate because of their real-time flexibility.

Adaptive beamforming is meant to decrease the expenses for 4G technology implementations by reducing the number of base and relay stations required.

By exploiting and improving the main advantages of digital beamforming methods and at the same time eliminating the disadvantages, these techniques may provide better solutions for the contemporary tasks in wireless communications, such as 4G communications systems, and other applications of signal processing.

REFERENCES

