A High-Speed Adaptive Antenna Array with Simultaneous Multiple-Beamforming Capability

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Abstract — **A new type of adaptive beamforming antenna system architecture is developed for multi-channel wireless communications. Multi-beam beamforming with high data throughput is accomplished using the proposed beamformer architecture consisting of analog mixers and multi-tone direct digital synthesizers (DDS). The multi-beam beamformer is based on multi-tone weighting scheme combined with analog-digital hybrid signal processing. High-speed real-time beamforming is realized by analog beamformer circuit, while the flexibility on adaptive beamforming algorithms is retained by computing weighting coefficients using the digital signal processor (DSP). A 5.8 GHz eight-element adaptive beamforming array successfully demonstrates two-beam simultaneous beamforming and two-channel data recovery at 25 Mb/s data throughput in each channel with BPSK modulation, based on SDMA (space division multiple access).**

I. Introduction

Adaptive beamforming arrays provide high flexibility in antenna pattern control and simultaneous multi-beam transmitting and receiving capability [1]. Conventional adaptive beamforming systems utilize a fully digital signal processing approach to accomplish the data modulation, coding, and adaptive beamforming [2]. However, when it comes to multi-beam beamforming, these systems suffer from the processing speed limit of the digital signal processors (DSP's) [3]. This is because of the heavy computation load on multi-channel throughput tasks including multi-beam beamforming and data modulation.

On the other hand, in analog beamforming systems combined with DSP for parallel weighting computation, high data throughput can be realized even in multi-channel signal processing since the tasks for real-time beamforming are essentially carried out not by DSP but by analog multipliers combined with heterodyne quadrature down-converters [4]. However, this analog-digital hybrid beamforming architecture requires a separate set of beamforming circuits for each beam synthesis [4]. Thus, it is unaffordable to implement this kind of system for multi-channel (or multi-beam) beamforming applications such as base station antennas where more than ten-beam outputs are required.

To develop a high-speed adaptive-beamforming smart antenna system with simultaneous multi-beam forming capability, we propose a novel array architecture using a combination of analog-digital hybrid beamforming and channel-level frequency division multiple access (FDMA). The crucial part of the system is a wideband multi-beam beamformer consisting of analog mixers and direct digital synthesizer (DDS). The DDS modulates weighting coefficient sets to multi-tone quadrature weightings. Thus, multi-beam beamforming for multiple channels is simultaneously performed by the analog mixers in real time in different frequency domains. After beamforming, those channels can be flexibly selected using band-pass filters. The selected channel signals are recovered by demodulation using corresponding multi-tone carriers.

In this way, not only is the beamformer reduced from M circuits to only one circuit for M-channel beam synthesis, but also the number of required analog mixers is reduced from ($M \times N$) to ($M + N$), where N is the number of antenna elements. In addition, the DDS can perform much faster signal generation speed than the DSP. Therefore, it is expected that the proposed multi-beam adaptive beamforming architecture can support high data throughput as well as rapid waveform generation capability, without great expense.

This paper is organized as follows. First, the system concepts on the proposed multi-beam beamformer are introduced. The measured component performances of the RF front-end and multi-beam beamformer are then given as a circuit overview. Finally, the system performance including two-channel adaptive beamforming and data recovery is presented as a validation of the original concept.

II. Multi-beam Beamformer Concept

Fig. 1 shows the block diagram of the proposed multi-beam adaptive-beamforming receiver array. It is assumed that the M-channel RF signals sharing a common RF carrier with different baseband data are incoming from different directions. First, the received M-channel RF

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signals in N element antenna array are down-converted to Low-IF signals. Then, the IF signals are split into two paths; one goes to the DSP controller and the other to analog mixers in the multi-beam beamformer. In the DSP controller, the IF signals are sampled at the A/D converter. (M × N) sets of weighting coefficients are subsequently computed at DSP based on sampled IF signals.

At this moment, in traditional multi-beam beamformer architecture, (M × N) sets of analog mixers are needed for vector weighting multiplication. However, in the proposed architecture, the same weighting function is accomplished at an expense of only (M + N) sets of analog mixers using FDMA concept.

The M-tone I/Q DDS plays an important role in this multiplexing process. The main role of the DDS is to modulate M-channel weighting coefficient sets onto M-tone carriers in quadrature phase modulation (QPSK) form. The time domain DDS output signal at n-th element is expressed as:

$$s_n(t) = \sum_{m=1}^{M} (Wr_{m,n} \cos \omega_m t + Wi_{m,n} \sin \omega_m t) \qquad (1)$$

where *Wr* and *Wi* represent real and imaginary components of complex weighting coefficients and ω_m is the carrier frequency for m-th channel.

In the multi-beam beamformer, the vector multiplication of the M-tone weightings for N elements from the DDS and N-element IF signals from the down-converter is essentially carried out by N analog mixers. Once this multiplication process is done at each mixer, the IF signals are not only weighted to form M independent beams, but also up-converted to M-channel frequencies. The weighting vector summation is subsequently carried out using an N-way power combiner where the multi-tone weighting process is completed.

After this weighting process, M sets of bandpass filters (BPFs) are used for channel selection. Then, the spectrally separated signals are down-converted by the corresponding carrier frequencies ω_m using M analog mixers. The IF signals are finally recovered throughout the lowpass filters where the unwanted harmonic components are removed. The baseband signals are obtained using any standard IF receiver in the subsequent stage.

Thus, the proposed multi-beam beamformer can reduce the number of mixers from (M × N) to (M + N) for weighting multiplication process. In addition, the down-conversion does not require to separate the IF signals into in-phase (I) and quadrature-phase (Q) signals because the DDS can generate the weighting in QPSK form. Since the I/Q down-converter and I/Q mixers for weighting process are not required, the number of mixers is further reduced by a factor of 2. This results in a simple and cost-effective down-converting and beamforming architecture for multi-beam beamforming applications.

The M-channel multi-beam beamformer can create M sets of independent beams pointing toward Signals of Interest (SOIs) by providing a maximum gain to each estimated Direction of Arrival (DOA). Meanwhile, the beamformer also suppresses the interference and multi-path signals by creating up to (M-1) sets of distinct nulls toward Signals Not of Interest (SNOIs).

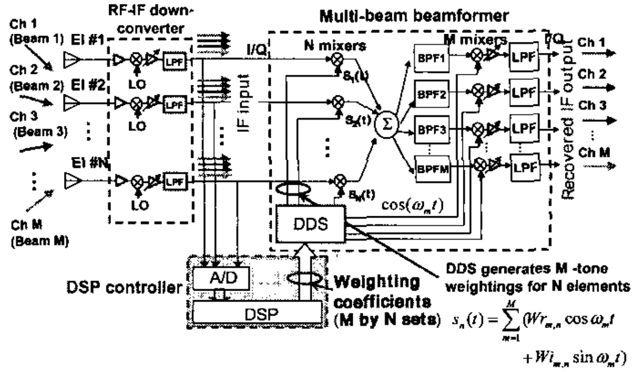

Fig. 1. Proposed multi-beam adaptive-beamforming receiver system block diagram.

III. CIRCUIT OVERVIEW

The proposed multi-beam adaptive-beamforming antenna system consists of a planar antenna array, RF-IF down-converter, and multi-beam beamformer. The test-bed with an eight-element linear array is capable of two-channel simultaneous beamforming and data recovery based on SDMA concept (M=2, N=8).

Fig. 2 shows the RF front-end consisting of a broad-band quasi-Yagi array [5] and an RF-IF down-converter.

The RF-IF down-converter consists of three-stage low noise amplifiers with 34.5 dB gain, RF mixers with 9.5 dB conversion loss at +10 dBm LO power, IF variable gain amplifiers (IF-VGAs) with up to 25 dB gain, and low pass filters. For −50 dBm RF input power at 5.8 GHz, the IF signal power is boosted up to 0 dBm at 30 MHz by the IF-VGAs. The IF-VGAs are also used to calibrate gain errors among elements at IF signals. The phase errors in the down-converter are also calibrated at IF signals before weighting computation to minimize the DOA estimation error.

Fig. 3 shows the multi-beam beamformer including the analog beamformer and two-tone I/Q DDS. The two-tone I/Q DDS architecture is constructed using 8-bit D/A converters, differential amplifiers, quadrature modulators, and bandpass filters. (2 × 8) sets of the computed weighting coefficients (for two channels, I/Q pair, and 8 elements) in the DSP are directly applied in digital form to the DDS, where two-tone quadrature weightings for eight elements are generated. Two-tone carrier frequencies are 230 MHz and 390 MHz, respectively. The recovery quadrature modulator is used to provide a synchronized LO source in channel recovery process. In preliminary measurement, the output amplitude linearity error is less than 0.5 % and the phase error is less than 3 degree in each element. The gain error between channels (230 MHz and 390 MHz) is less than 0.5 dB and the phase error among elements is less than 10 degree. Thus, overall scan angle error caused by weighting process at the DDS is negligibly small. Although the DDS architecture in this prototype is constructed using a number of components, the whole structure can be integrated into one chip in current DDS technology.

The analog beamformer mainly consists of weighting mixers (for up-conversion), channel-selection BPFs, and recovery mixers (for down-conversion).

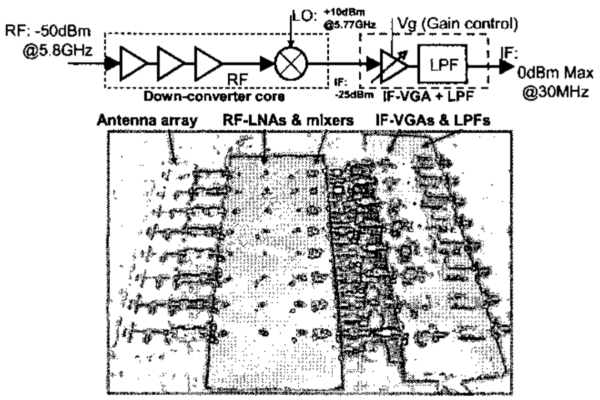

Fig. 2. RF-front-end.

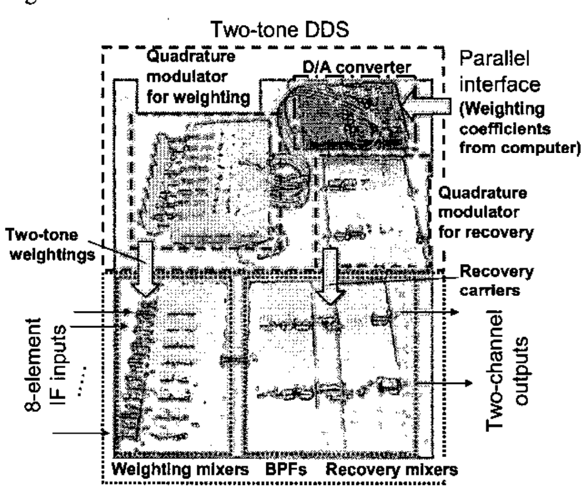

Fig. 3. Multi-beam beamformer.

In the experiment, the A/D converter for sampling and the DSP for weighting computation are replaced by eight-channel digital oscilloscope with 200 Ms/s sample rate and the personal computer, respectively. The sampled IF signals at the digital oscilloscope are digitally transferred to the computer throughout the GP-IB interface. The DSP function is carried out by MATLAB. The weighting computation in MATLAB is implemented in the following manner. The quadrature components of the sampled 8-element IF signals are first obtained using the Hilbert transform. The DOA estimation is then carried out using one of the standard algorithms, ESPRIT [6]. Finally, the Sample Matrix Inversion (SMI) algorithm [7] is used to compute complex weighting coefficients for main-lobe steering toward SOIs and for nulling toward SNOIs. The computed weighting coefficient sets are digitally transferred from the computer to the D/A converter in the DDS throughout the parallel interface.

IV. SYSTEM PERFORMANCE

A. SDMA performance in two-beam simultaneous beamforming

The purpose of this experiment is to perform the SDMA capability of the proposed multi-beam beamforming architecture. In this experiment, two-beam simultaneous beamforming based on SDMA has been tested using the 8-element array test-bed. The measurement is carried out in the open space of the conference room where strong reflections from the ceiling, floor, and walls surrounding the receiver are expected. Two unmodulated RF transmitters are respectively located at +36° (source 1) and 0° (source 2) in the same distance from the test-bed receiver.

First, the signal DOAs are estimated using ESPRIT algorithm from the sampled 8-element IF signals. In the above source locations, The DOA estimations are +37.0° and –0.6° so that the DOA estimation errors are +1° and –0.6° for source 1 and source 2, respectively.

Fig. 4 shows the measured beamforming results. The solid lines represent the beamforming patterns extracted from the weighted two-tone outputs in channel 1 (Ch1) and channel 2 (Ch2) of the multi-beam beamformer. The dashed lines are the estimated patterns from the computed weighting coefficients based on the DOA estimations. The channel 1 output forms a pattern with a peak pointing toward source 1 direction, and a null toward source 2 direction. Meanwhile, the channel 2 output steers a peak toward source 2 direction and creates a null toward source 1 direction. The interference suppression from peak to null is around 20 dB for both channels in measured patterns.

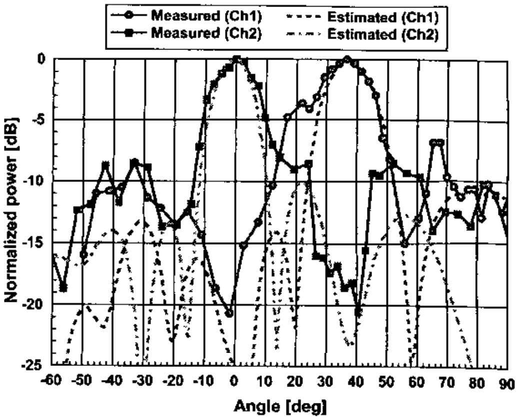

Fig.4. Patterns of two-beam simultaneous beamforming.

B. Two-channel data recovery performance

A measurement of the two-channel simultaneous data recovery is carried out using the two RF transmitters positioned in the same locations as described in the previous section (IV-A), i.e. source 1 at + 36° and source 2 at 0°. In this case, two RF source signals are modulated by 25 Mb/s BPSK with a common RF carrier at 5.8 GHz and are simultaneously transmitted. Different baseband data with the same data rate are used for source 1 and source 2 to demonstrate the two-channel simultaneous data recovery. The baseband data for source 1 is a 3-bit pseudo noise (PN) random sequence code, and the one for source 2 is a periodic binary code.

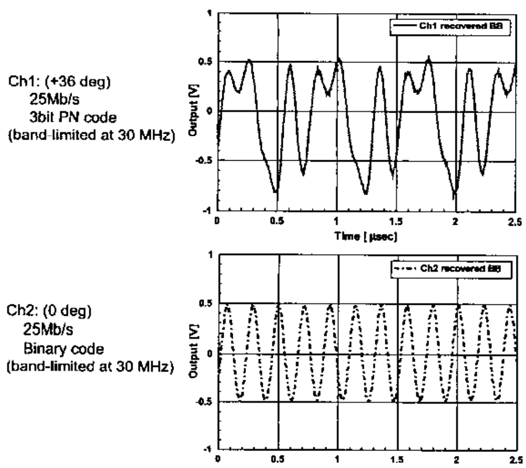

Fig. 5. Two-channel simultaneous data recovery.

Fig. 5 shows the simultaneously recovered time-domain baseband data waveforms from two channel outputs of the multi-beam beamformer. The baseband data are properly extracted with slight signal distortions due to the bandwidth limitation of the 30 MHz low-pass filters placed on the output stage of the multi-beam beamformer. This validates successful multi-channel data recovery based on the SDMA concept of the proposed multi-beam adaptive-beamforming smart antenna system.

V. CONCLUSION

A new type of the multi-beam adaptive-beamforming smart antenna system test bed has been developed. The analog-digital hybrid beamforming approach is combined with channel-level FDMA mux/demux concept to accomplish high-speed real-time multi-beam beamforming for multi-channel applications. The 5.8 GHz test-bed successfully demonstrates the two-beam simultaneous beamforming within 1° DOA estimation error. The interference suppression is 20 dB. The data throughput of 25 Mb/s in each channel has been demonstrated in two-channel BPSK data recovery test for SDMA capability. The same architecture can be used to increase more channel capacity with much higher data rate on the order of hundreds of Mb/s in current 5.8 GHz system. This new multi-beam beamforming smart antenna architecture ensures a low-cost, high-speed, real-time adaptive beamforming and data recovery solution for multi-channel wireless communications with SDMA concept.

ACKNOWLEDGEMENT

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