Design by Mathematics of Full Software Radio circuits and systems: methodology and application to 5G standard

Organizers
François Rivet, University of Bordeaux, France

Abstract
The goal of this tutorial is to present our original methodology of Full Software Radio system design. The diversity of communication standards implies the use of multi-band and multi-mode radios. That is why Full Software Radio proposes to challenge a new way of integrating RF circuits and systems by tackling the main issue: transceiving concurrently any RF signal within a very wide band of interest for telecommunication industry, from DC to 5 GHz for instance. The focus of this tutorial will concern the design by mathematics of such RF transceiver, exploring novel approaches along with a thorough discussion of advanced techniques for these receivers and transmitters towards a revolution in RF integrated circuits and systems using 28 nm FDSOI STMicroelectronics technology.

Programme

14:20 - 14:30 Welcome

14:30 - 16:00 Sampled Analog Signal Processor (SASP)
François Rivet, University of Bordeaux, France
→ Abstract

16:00 - 16:30 Coffee Break

16:30 - 17:15 Walsh transmitter
Yann Deval, University of Bordeaux, France
→ Abstract

17:15 - 18:00 The Riemann Pump
Yoan Veyrac, University of Bordeaux, France
→ Abstract

18:00 - 18:10 Discussion
14:30 - 16:00  Sampled Analog Signal Processor (SASP)
François Rivet, University of Bordeaux, France

Abstract
SASP Rx is a frequency domain receiver. The principle of the SASP aims at selecting a spectral envelope of a RF signal within a very wide frequency band. To reach this target, the SASP processes analogically the RF input signal spectrum thanks to an analog Discrete Time Fourier Transform (DFT) with discrete time voltage samples. Once the spectrum is processed, voltage samples representing the spectral signal envelope to be treated are converted into digital. The selection of few voltage samples among thousands replaces the classical mixing and filtering operations. It reduces the A/D conversion frequency from GHz frequencies to MHz ones and thus allows a multi band selection at a very low power consumption. The purpose is to extract the desired input signal spectrum envelope by recovering its spectrum. The RF signal is sampled and transformed. Among thousands of voltage samples, only the ones representing the RF signal envelope are sent out towards an A/D converter. The selection is not limited to only one voltage sample representing one band but can be extended concurrently to several bands.

16:30 - 17:15  Walsh transmitter
Yann Deval, University of Bordeaux, France

Abstract
Walsh is a frequency domain transmitter. According to the Fourier theory, any signal can be decomposed into a series of harmonics. For example, a sum of sine waves can generate a square wave. A harmonic is a sine wave at a particular frequency. A sine wave at a particular frequency can be generated by a Voltage Controlled Oscillator (VCO). However, the analog implementation within a single chip of a given number of VCOs to generate a wide variety of harmonics is impossible because of die area, cross-talk, pairing and power consumption reasons. The solution to this paradigm is to generate a very high frequency and to divide it by 2 a given number N of times. The division is easily integrated into silicon (flip-flop) with the difference that it generates square signals. A family of square signals is generated. As proposed here, square signals can be generated from a high frequency and divide by 2, N times thanks to a mm-wave Phase-Locked Loop (PLL). It is a matter of recombining spectrum by eliminating undesired harmonics with existing ones in square signal family. The mathematical theory is based on Walsh-Fourier theorem which demonstrates that a family of square waves can generate any kind of signal. This theorem clearly states that intermediate signal coming from the mm-wave-PLL are sufficient using algebraic operations (phase shifting, sum, delay ...) to implement this signal generator. Once the signals are generated, they must be amplified. The originality of this project is that every square signals are directly amplified and then combined by their power to form the transmitted signal with a correct matching. The sum is performed thanks to a current node. Algebraic operations are consequently carried out by biasing differential amplifying square signals.

17:15 - 18:00  The Riemann Pump
Yoan Veyrac, University of Bordeaux, France

Abstract
Riemann Pump is a time-domain transmitter. The purpose of Riemann Pump is to generate arbitrary waveforms up to the GHz range with a low cost and low consumption solution, the main target being the generation of modulated signals. At first, the wanted signal is available in a high-resolution digital representation. This signal is converted into a lighter size with a specific differential coding, and then converted in the analog domain thanks to a suited DAC. An analog version of the wanted signal is thus generated thanks to a pre-determined set of slopes. A Digital Signal Processor (DSP) computes the Riemann code (i.e. the slopes index sequence) from the theoretical desired signal. This code controls switched current sources, in order to produce current steps that are integrated into an output capacitive load, producing a piecewise linear approximation of the wanted signal. The current sources and the capacity of the load are calibrated with respect to the wanted bandwidth and dynamic. The digital to analog (DA) operation consists in pumping charges into a capacitor to generate the wanted signal. It involves integration over time to approximate this signal, reminding us of the Riemann Integral. The developed circuit is named the Riemann Pump, in reference to this founding principle.