# Adaptive Polynomial Transversal Filtering

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## Introduction

The scaling down of CMOS technology is brought up against analog circuit design, and settling these problems is a challenging prospective research area. The adaptive polynomial transversal filter (APTF) based on the least-mean square (LMS) algorithm is one promising digital solution to compensate for the universal errors in discrete analog systems. However, the large computation and iteration demand long training time and hamper the efficiency of the simulation-based empirical analysis. With the help of GPU acceleration, we expect to speed up the simulation without losing accuracy.

## Description

The APTF simulation module in Fig. 1 is largely divided into three parts, error generation, calibration, and analysis. Since the circuit model represents nonlinear errors by using recursive polynomial functions, both the error generation and calibration blocks include many multiplier and adder arrays. Another major feature of these blocks is the high data dependency of codes to handle the memory effect from feedback circuits. It would be the most critical bottleneck in parallel processing. Fig. 2 shows the conceptual diagram of APTF and LMS blocks in the error calibration part. In the error analysis, the FFT function is repeatedly called, which can be easily accelerated by a GPU implementation.

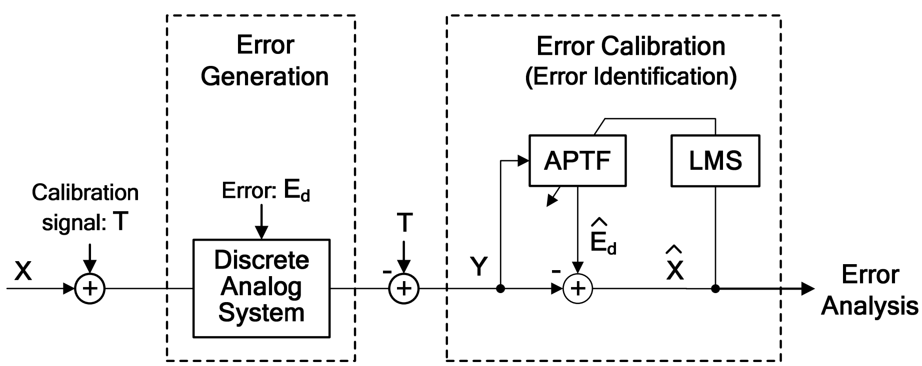
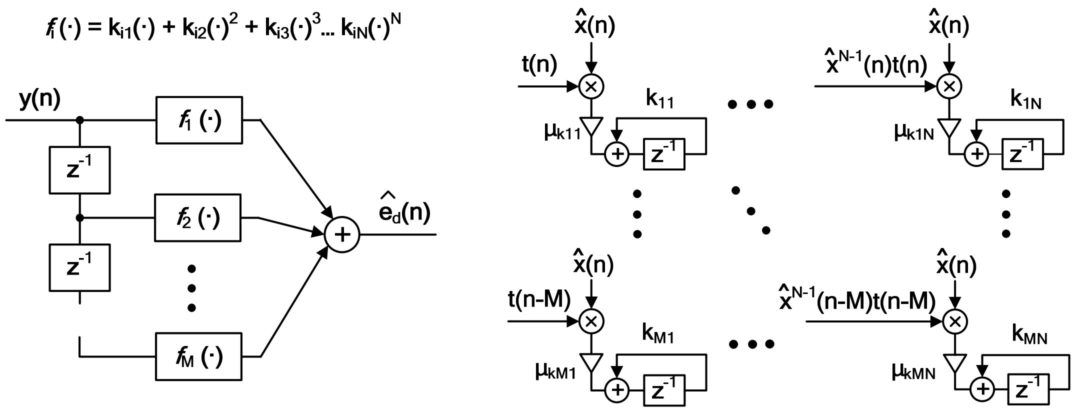



Figure 1. Block diagram of the APTF module.

For the purpose of an efficient implementation, previously developed MATLAB modules (Fig. 1) will be provided, to be accelerated by GPU through this project.




(a) (b)

Figure 2. (a) APTF and (b) LMS blocks.

## Objective

Through the proper partitioning and optimization of GPU implementation, we expect to speed up the convergence of the APTF simulation to hours, than days currently executed on a dual-core desktop computer.

## Background

A background in digital and analog signal processing is helpful. Especially, the understanding of adaptive filtering and basic amplifier design will be very helpful.

## Resources

The book, “Digital Signal Processing” written by J. Proakis and D. Manolakis (Prentice Hall, 2007), is recommended for adaptive filtering. There are many online tutorials and lectures about adaptive filtering, including
http://www.eece.unm.edu/faculty/bsanthan/ADAP/adap.htm,
http://homes.esat.kuleuven.be/~sszczepa/dspII/2008-2009/index.html - Slides, etc.

In terms of software, there are multiple adaptive filter implementations and analog/mixed-signal circuit models available in
http://www.mathworks.com/matlabcentral/fileexchange/.

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