

COURSE DESCRIPTION

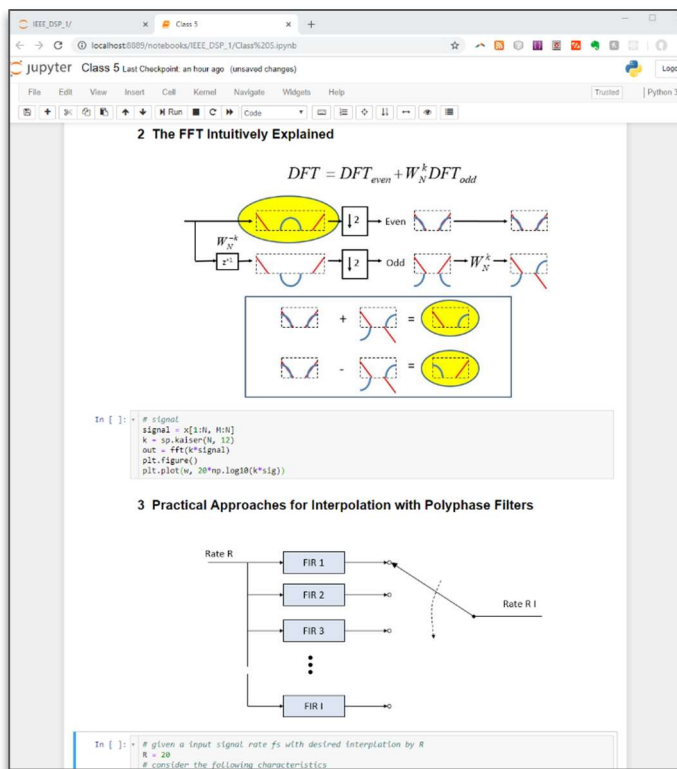
Course Name: DSP For Wireless Communications
Course Start Date: Oct 14, 2021, videos released weekly 2x1.5 hours
Workshops: Thursdays: Oct 21, 28, Nov 4, 11, 18 7-8pm EST
Location: Zoom Webinar
Speaker: Dan Boschen

New Format Combining Live Workshops with Pre-recorded Video

This is a hands-on course providing pre-recorded lectures that students can watch **on their own schedule** and **an unlimited number of times** prior to live Q&A/Workshop sessions with the instructor. Ten 1.5 hour videos released 2 per week while the course is in session will be available for up to two months after the conclusion of the course.

Course Summary

This course is a fresh view of the fundamental and practical concepts of digital signal processing applicable to the design of mixed signal design with A/D conversion, digital filters, operations with the FFT, and multi-rate signal processing. This course will build an **intuitive** understanding of the underlying mathematics through the use of graphics, visual demonstrations, and applications in GPS and mixed signal (analog/digital) modern transceivers. This course is applicable to DSP algorithm development with a focus on meeting practical hardware development challenges in both the analog and digital domains, and not a tutorial on working with specific DSP processor hardware.



The screenshot shows a Jupyter Notebook titled "Class 5 Last Checkpoint: an hour ago (unsaved changes)". The notebook content includes:

- 2 The FFT Intuitively Explained**: A diagram illustrating the decomposition of the DFT into even and odd components. The equation $DFT = DFT_{\text{even}} + W_N^k DFT_{\text{odd}}$ is shown. The diagram shows a signal $x[n]$ being split into even and odd parts, each processed by a filter (labeled "Even" and "Odd"), and then combined. Below this, two equations show the addition and subtraction of signals: $x[n] + x[n+N/2] = 2 \cos(\pi n/2) x_1[n]$ and $x[n] - x[n+N/2] = 2j \sin(\pi n/2) x_2[n]$.
- 3 Practical Approaches for Interpolation with Polyphase Filters**: A block diagram showing a signal at Rate R being processed by a series of FIR filters (FIR 1, FIR 2, FIR 3, ..., FIR I) to produce a signal at Rate R I.
- Code cells: The first code cell defines a signal $x[n]$ and computes its FFT. The second code cell defines a function for interpolation using polyphase filters.

Now with Jupyter Notebooks!

This long-running course has been updated to include Jupyter Notebooks which incorporates graphics together with **Python simulation code** to provide a "take-it-with-you" interactive user experience. No knowledge of Python is required but the notebooks will provide a basic framework for proceeding with further signal processing development using that tools for those that have interest in doing so.

This course will not be teaching Python, but using it for demonstration. A more detailed course on Python itself is covered in a separate course also taught by Dan titled "Python Applications for Digital Design and Signal Processing".

Students will be encouraged but not required to load all the Python tools needed, and all set-up information for installation will be provided prior to the start of class.

Target Audience:

All engineers involved in or interested in signal processing applications. Engineers with significant experience with DSP will also appreciate this opportunity for an in-depth review of the fundamental DSP concepts from a different perspective than that given in a traditional introductory DSP course.

Benefits of Attending/ Goals of Course:

Attendees will build a stronger intuitive understanding of the fundamental signal processing concepts involved with digital filtering and mixed signal analog and digital design. With this, attendees will be able to implement more creative and efficient signal processing architectures in both the analog and digital domains. The knowledge gained from this course will have immediate practical value for any work in the signal processing field.

Topics / Schedule:

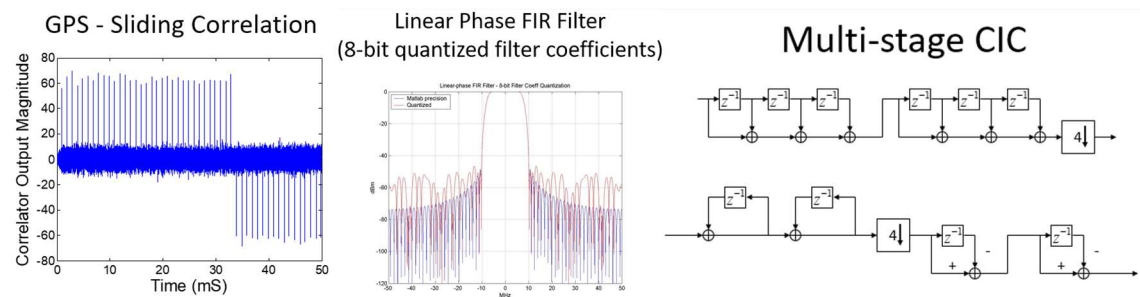
Class 1: Correlation, Fourier Transform, Laplace Transform

Class 2: Sampling and A/D Conversion, Z –transform, D/A Conversion

Class 3: IIR and FIR Digital filters, Direct Fourier Transform

Class 4: Windowing, Digital Filter Design, Fixed Point vs Floating Point

Class 5: Fast Fourier Transform, Multi-rate Signal Processing, Multi-rate Filters



Speaker's Bio:

Dan Boschen has a MS in Communications and Signal Processing from Northeastern University, with over 25 years of experience in system and hardware design for radio transceivers and modems. He has held various positions at Signal Technologies, MITRE, Airvana and Hittite Microwave designing and developing transceiver hardware from baseband to antenna for wireless communications systems. Dan is currently at Microchip (formerly Microsemi and Symmetricom) leading design efforts for advanced frequency and time solutions.

For more background information, please view Dan's Linked-In page at: <http://www.linkedin.com/in/danboschen>