Monson H. Hayes and John Treichler

Adaptive Filtering

Please send suggestions for web resources of interest to our readers, proposals for columns, as well as general feedback, by email to Alen Docef ("Best of the Web" associate editor) at adocef@vcu.edu.

In this issue, "Best of the Web" focuses on adaptive filtering or, more generally, adaptive signal processing–the design of time-varying (adaptive) digital filters that would tune themselves to optimally process nonstationary signals in nonstationary environments.

Much of what is found today in adaptive filtering algorithms can be traced back to two seminal articles that were published in 1960. The first article, "Adaptive Switching Circuits," was published by Bernard Widrow and Marcian Hoff, and described the least mean square (LMS) algorithm. This algorithm is widely used in adaptive signal processing, and is the most well-understood approach to training a linear system to minimize the mean square error. Appearing in 1960, the second article, "A New Approach to Linear Filtering and Prediction Problems," was authored by R.E. Kalman and described a recursive solution to the discrete-data linear filtering problem. Since that time, the Kalman filter has been the subject of extensive research and application.

The area of adaptive signal processing has had a significant impact on a wide variety of signal processing applications. These include inverse filtering, signal modeling, prediction, channel equalization, echo cancellation, noise cancellation, system identification and control, line enhancement, adaptive notch filtering, and beam forming. These developments have been crucial to the practical and commercial success of many types of communications systems. Without adaptive DSP we'd still be communicating at 1,200 b/s.

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For the resources presented here, which are by no means exhaustive, the attributes in square brackets describe the type of information that is contained in the given resource. Unless otherwise noted, the resources are free. This resource list is also available by convenient point and click on the *IEEE Signal Processing Magazine* Web site at http://apollo.ee.columbia.edu/spm/?i=external/bow.

ADAPTIVE FILTERING TUTORIALS AND OVERVIEWS

WIKIPEDIA

http://en.wikipedia.org/wiki/Adaptive_filter
http://en.wikipedia.org/wiki/Least_mean_squares_filter
http://en.wikipedia.org/wiki/Recursive_least_squares_filter
http://en.wikipedia.org/wiki/Kalman_filter
[tutorial]

A number of tutorials on the key adaptive filtering algorithms may be found at Wikipedia, and some of these tutorials provide a number of useful links to additional papers and information. These articles are good starting points for an introductory overview of adaptive signal processing.

CONNEXIONS

http://cnx.org/content/col10280/latest/
[tutorial]
http://cnx.org/content/m11433/latest/
(Introduction to Adaptive Filtering)
http://cnx.org/content/m11801/latest/
(Adaptive FIR Filtering Laboratory)
http://cnx.org/content/m11438/latest/
(Kalman Filtering)
http://cnx.org/content/m11437/latest/
(Kalman Filtering Applications)
[modules]

Connexions is an environment for collaboratively developing, freely sharing, and rapidly publishing scholarly content on the Web. The educational materials are organized in small modules, and are the basic building blocks of a Connexions course, textbook, or other type of collection. These modules represent a small knowledge chunk that contains text and images that address a single topic or a specific aspect of a topic.

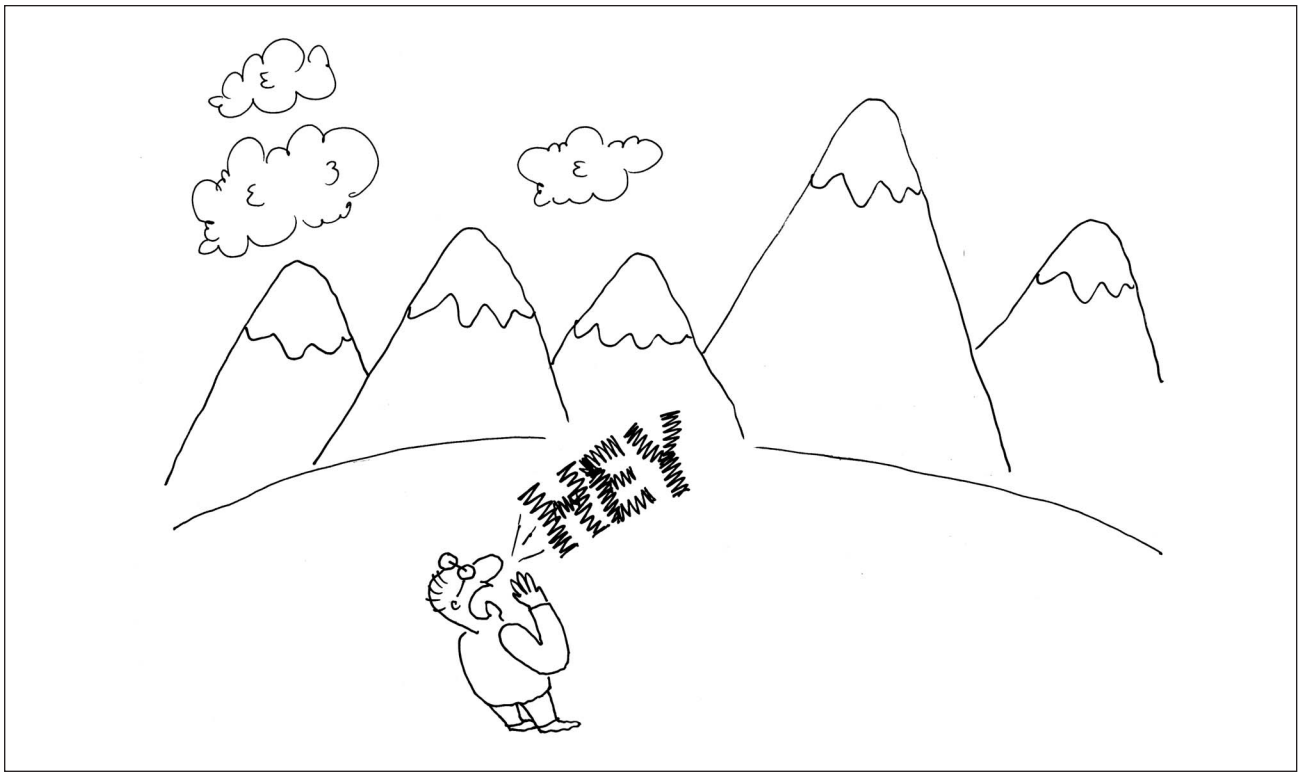
A course that has been assembled by Douglas Jones from modules he created is available from the resource listed above. This course covers different types of adaptive filters and their implementation, with an in depth analysis for the Wiener Filter algorithm and the LMS algorithm.

Some additional modules not included in this course, but relevant to adaptive signal processing, are accessible using the last four links.

KALMAN FILTER PORTAL

http://www.cs.unc.edu/~welch/kalman/
http://www.cs.unc.edu/~tracker/ref/s2001/kalman/index.html
[one-stop information portal]

An excellent and extensive one-stop information portal for a wide variety of resources related to the Kalman filter may be found at the site maintained by Greg Welch and Gary Bishop at the University of North Carolina at Chapel Hill. Included on this portal are links to tutorials, papers, references, software, courses, a Java-based learning tool, and research related to the Kalman filter.



Cartoon by Tayfun Akgul (tayfun.akgul@ieee.org).

Accessible from the Kalman filter portal are course materials for an excellent SIGGRAPH 2001 course "An Introduction to the Kalman Filter," by Greg Welch and Gary Bishop. This course is a stand-alone reference that is valuable to anyone beginning to learn about the Kalman filter.

ADAPTIVE SIGNAL PROCESSING COURSE

http://homes.esat.kuleuwen.be/~moonen/asp_course.html
[tutorial]

This resource contains seven chapters from a book on adaptive filtering by Marc Moonen and Ian Proudler that are used for a course on adaptive signal processing. The chapters cover Wiener filtering and least squares estimation, recursive least squares (RLS), the LMS algorithm, and the Kalman filter.

SOFTWARE AND DEMOS

ADAPTIVE FILTERING DEMOS IN MATLAB

http://www.mathworks.com/matlabcentral/fileexchange/loadFile.do?objectId=9259&objectType=file
[software]

A very good collection of demos on adaptive filtering has been assembled by Ricardo Losada. These demos use various adaptive filtering algorithms such as LMS, RLS, normalized least mean square (NLMS), frequency-domain adaptive filters, filtered-X LMS, and block LMS, and include applications in adaptive channel equalization, adaptive noise canceling, linear prediction, active noise control, adaptive line enhancement, and acoustic echo cancellation. The downloadable zip file includes well-documented Matlab code and detailed descriptions of the demos in PDF format. Each Matlab file, along with the accompanying documentation, provides a self-contained demo that is easy to use and simple to modify. Several of the demos include processing of real audio files.

ADAPTIVE SIGNAL PROCESSING JAVA TEACHING TOOL

http://www.eee.strath.ac.uk/r.w.stewart/adaptivejava/begin.htm
[software]

This resource provides an overview of adaptive filtering and a link to an excellent Java applet illustrating the use of adaptive filters for system identification. In this applet, the system that is to be identified may be either finite-impulse response (FIR) or infinite-impulse response (IIR), or it may be a system specified by the user. The applet allows for eight different FIR adaptive filtering algorithms for system identification including LMS, least squares with QR decomposition, and RLS. It also has four different IIR adaptive algorithms: the IIR LMS algorithm, the simplified gradient algorithm, the full gradient algorithm, and the QR pseudo linear regression algorithm. A nice feature is that it allows a side-by-side comparison of two adaptive filtering algorithms that may be selected by the user.

ADAPTIVE FILTERING IN JAVA

http://www.developer.com/java/other/article.php/3549991 (Getting Started)
http://www.developer.com/java/other/article.php/3560501 (Adaptive Whitening Filter)
http://www.developer.com/java/other/article.php/3566951 (LMS)

http://www.developer.com/java/other/article.php/3576061
(Adaptive Line Tracking)
http://www.developer.com/java/other/article.php/3583241
(Adaptive Identification and Inverse Filtering)
http://www.developer.com/java/other/article.php/3599661
(Adaptive Noise Cancellation)
http://www.developer.com/java/other/article.php/3610806
(Adaptive Prediction)
[software]

These resources are a set of lessons designed to teach one how to do adaptive filtering in Java. The first lesson, "Adaptive Filtering in Java, Getting Started," introduces the topic of adaptive filtering by showing how to write a program in Java to adaptively design a time-delay convolution filter with a flat amplitude response and a linear phase response using the LMS algorithm. The second lesson shows how to write an adaptive whitening filter program in Java to extract wide band signal from a channel in which the signal is corrupted by one or more components of narrow band noise. The third lesson presents a general-purpose LMS adaptive engine written in Java that can be used to solve a wide variety of adaptive filtering problems. Following this, the remaining four lessons are concerned with four common applications in which adaptive filtering is used: line tracking, identification and inverse filtering, noise cancellation, and prediction.

KALMAN FILTERING TOOLBOX

http://www.cs.ubc.ca/~murphyk/Software/Kalman/kalman.html
[software]

A Kalman filter Web page that provides a Kalman filtering toolbox in Matlab (including demos) was created by Kevin Murphy at the University of British Columbia. The page also includes a short overview of the Kalman filter.

ADAPTIVE IIR FILTERING IN MATLAB AND OCTAVE

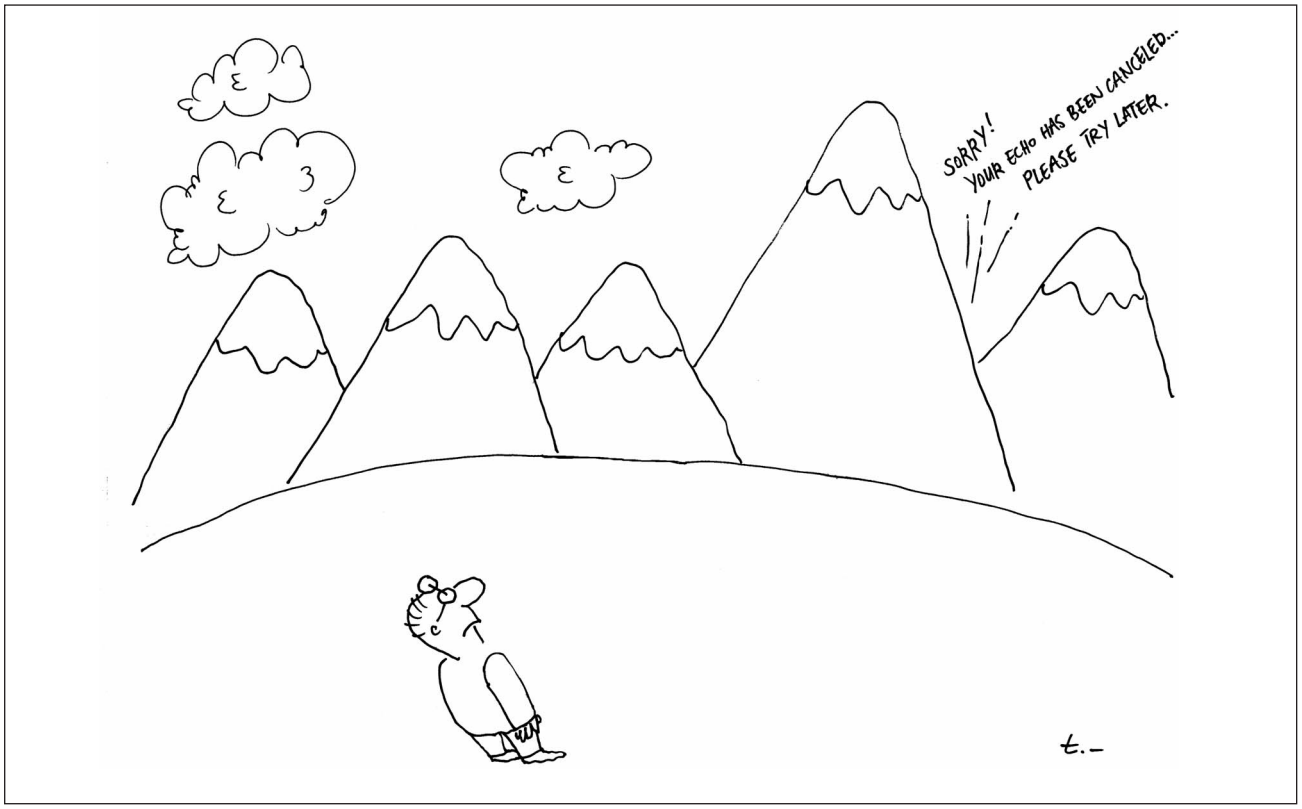
http://www-public.it sudparis.eu/~regalia/adapiir/
[software]

Philip Regalia, the author of the seminal textbook of adaptive IIR filters, *Adaptive IIR Filtering in Signal Processing and Control*, maintains a Web page that contains Matlab m-files for adaptive IIR filtering. Since adaptive IIR filter design is much more difficult and involved than adaptive FIR design, these m-files provide a useful and valuable resource. The m-files are upgraded versions of those that appear in his textbook and all have been tested with Matlab and Octave. The Web page also includes demos that illustrate the performance of the various Matlab functions provided in the package.

APPLICATIONS

ECHO CANCELLATION

http://en.wikipedia.org/wiki/Echo_cancellation
http://www.iec.org/online/tutorials/echo_cancel/
http://users.ece.gatech.edu/~barry/digital/supp/20echo.pdf
http://www.commsdesign.com/showArticle.jhtml?articleID=16502325
(Part 1)



Cartoon by Tayfun Akgul (tayfun.akgul@ieee.org).

http://www.commsdesign.com/design_corner/OEG20031009S0005 (Part 2)
http://www.itu.int/rec/T-REC-G.168/e [tutorial]

Echo cancellation is an important application of adaptive filtering and research continues to make echo cancellers more effective in environments that, in some cases, may be harsh or complex. Listed above are many excellent tutorial reviews of echo cancellation.

The first is a concise overview of echo cancellation in Wikipedia that contains additional links. The second is an online tutorial from the International Engineering Consortium. For registered members (registration is free), a complete PDF version of the tutorial may be downloaded. The third is a chapter from a book by John Barry from the Georgia Institute of Technology that provides a nice overview of echo cancellation, along with some exercises and problems. The last is a two part series on echo cancellation from CommsDesign. The first part examines the basic causes of echo in a networking design and provides detailed insight into the echo cancellation architectures that can be used to solve these echo problems. The second part looks at some of the implementation challenges designers will face and some solutions to the problems.

The last link above points to a valuable resource for implementers of digital communications systems over the public switched telephone network, i.e., the ITU-T G.168 Standard, *Digital Network Echo Cancellers*. This standard describes characteristics and requirements for line echo cancellers that are part of such systems.

ADAPTIVE NOISE CANCELLATION

http://www.owlnet.rice.edu/~ryanking/elec431/
http://homes.esat.kuleuwen.be/~rombouts/demo/aec.html
https://gilbert.med.kuleuwen.be/~koen/demo_beam/demo_beam.html [demo]

The resources included here point to demos of adaptive noise cancellation. The first is a project on adaptive noise cancellation using the LMS and RLS algorithms. In addition to an overview of the problem of noise cancellation, and the LMS and RLS algorithms, the project team members provide Matlab m-files for the experiment. There is also an audio tour of the project that demonstrates the effects of different tap length and noise levels. Although the project setup is simple and idealized, it provides a nice introduction to the problem.

The second is a Web site that provides a demonstration of echo cancellation using several adaptive filtering algorithms: the normalized LMS algorithm, RLS, the fast Newton transversal filter, and the affine projection algorithm. The third provides an experiment in using beamforming with a microphone array for the enhancement of speech in a hands-free teleconferencing environment.

NOISE CANCELLATION AND ADAPTIVE TRACKING

http://www.eas.asu.edu/~dsp/grad/anand/demos.html [demo]

Arizona State University maintains a set of Java applets that provide demos of adaptive tracking of a time-varying system, adaptive noise cancellation, and different adaptive noise cancellers for audio. In addition to the Java applets, this Web page contains a brief description of the demo and the algorithms that are used.

AUTHORS

Monson H. Hayes (mhh3@gatech.edu) is a professor and associate chair of electrical and computer engineering at the Georgia Institute of Technology. He is a Fellow of the IEEE.

John Treichler (jrt@appsig.com) is the chief technical officer of Applied Signal Technology, Inc. and a Fellow of the IEEE. SP