ADAPTIVE FILTERS
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SUMMARY

In this up-to-date state of the art book, the authors provide a coherent and comprehensive introduction to adaptive filtering. They cover basic theory, practical realizations, and applications, such as adaptive equalizers for telecommunications data transmission systems. Practical engineers find this book a good source of information on the practical possibilities of these processors.

This book's key features include:

- Chapter 2 estimation theory discusses and is followed by two chapters on adaptive finite impulse response and infinite impulse response.
- Chapter 5 covers the theory, design, and application of adaptive lattice filters.
- Chapter 6 deals with signal transformation techniques for adaptive filtering.
- Chapter 7 covers adaptive filter implementations.
- Chapter 8 includes main applications in communications equalization and echo cancellation.

Chapter 9 describes such application areas as fast tracking filters for HF and microwave digital radion, linear predictive coding, and maximum-entropy and maximum-likelihood analysis techniques.

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Applications of adaptive filters include multichannel noise reduction, radar/sonar signal processing, channel equalization for cellular mobile phones, echo cancellation, and low delay speech coding. This chapter begins with a study of the state-space Kalman filter. In Kalman theory a state equation models the dynamics of the signal generation process, and an observation equation models the channel distortion and additive noise. Then we consider recursive least square (RLS) error adaptive filters. General discussion on how adaptive filters work, list of adaptive filter algorithms in DSP System Toolbox, convergence performance, and details on few common applications. Overview of Adaptive Filters and Applications. On this page. Adaptive Filters in DSP System Toolbox. Least Mean Squares (LMS) Based FIR Adaptive Filters. Recursive Least Squares (RLS) Based FIR Adaptive Filters. Affine Projection (AP) FIR Adaptive Filters. FIR Adaptive Filters in the Frequency Domain (FD).
Ein adaptiver Filter in der digitalen Signalverarbeitung ist ein spezielles digitales Filter, das die Eigenschaft besitzt, seine Übertragungsfunktion im Betrieb selbständig zu verändern. Blockdiagramm eines adaptiven Filters.

Deutsch Wikipedia: adaptive filter — adaptyvusis filtras

Savvybä—s dinamiųjai priklauso nuo filtruojamų duomenų specifikos. Preface. DSP and adaptive filtering. With the decrease in cost and the increase in speed of digital devices, Digital Signal Processing (DSP) is showing up in everything from cell phones to hearing aids to rock concerts. Many applications of DSP are static. More general adaptive filtering considerations. Following that, the lessons in the series will become somewhat more general. I plan to publish lessons that explain and provide examples for the four common scenarios in which adaptive filtering is used. Applications of Adaptive Filtering. By J. Gerardo Avalos, Juan C. Sanchez and Jose Velazquez. Submitted: October 15th 2010Reviewed: March 10th 2011Published: July 5th 2011.

The efficiency of the adaptive filters mainly depends on the design technique used and the algorithm of adaptation.