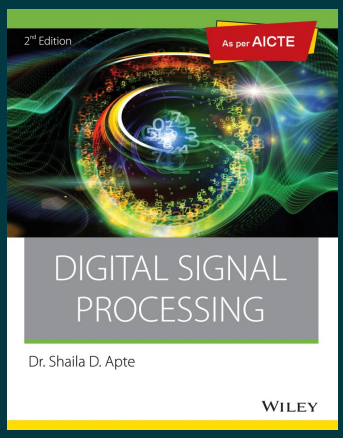
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Digital Signal Processing, 2ed, As per AICTE

By Dr. Shaila D. Apte

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Description

The scope of the book has been increased to cover the syllabi of DSP course all over India and to enhance the practical guidance in the book by including the MATLAB programs for teachers teaching the subject for the first time. The efforts are made to enhance the scope of the book to its fullest possible extent by and to include large number of MATLAB programs for the benefit of the reader. The book is intended to provide rigorous treatment of DSP at undergraduate level and will serve as a textbook for undergraduate studies and is designed to provide solid foundation for specialized courses in DSP.

About the Author

Dr. Shaila D. Apte

Dr. Shaila Dinkar Apte is currently working as a Professor in Rajarshi Shahu College of Engineering, Pune and as a reviewer for the International Journal of Speech Technology by Springer Publication, International Journal of Digital Signal Processing, Elsevier Publication and Bulletin of Pure and Applied Mathematics (BPAM). She is currently guiding 8 PhD candidates and about 23 candidates completed their M.E. dissertations under her guidance. With a vast teaching experience of 28 years in Electronics Engineering, she enjoys great popularity amongst students. She has been teaching Digital Signal Processing for the last 18 years.

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