

1. Record Nr.	UNISA996466285403316
Titolo	Multi-disciplinary Trends in Artificial Intelligence [[electronic resource]] : 12th International Conference, MIWAI 2018, Hanoi, Vietnam, November 18–20, 2018, Proceedings // edited by Manasawee Kaenampornpan, Rainer Malaka, Duc Dung Nguyen, Nicolas Schwind
Pubbl/distr/stampa	Cham : , : Springer International Publishing : , : Imprint : Springer, , 2018
ISBN	3-030-03014-8
Edizione	[1st ed. 2018.]
Descrizione fisica	1 online resource (XV, 271 p. 106 illus., 73 illus. in color.)
Collana	Lecture Notes in Artificial Intelligence ; ; 11248
Disciplina	006.3
Soggetti	Artificial intelligence Computer communication systems Special purpose computers Optical data processing Algorithms Computers Artificial Intelligence Computer Communication Networks Special Purpose and Application-Based Systems Computer Imaging, Vision, Pattern Recognition and Graphics Algorithm Analysis and Problem Complexity Information Systems and Communication Service
Lingua di pubblicazione	Inglese
Formato	Materiale a stampa
Livello bibliografico	Monografia
Nota di contenuto	Artificial intelligence -- Natural language processing -- Knowledge representation and reasoning -- Planning and scheduling Computer vision -- Machine learning -- Modeling and simulation -- Distributed artificial intelligence.
Sommario/riassunto	This book constitutes the refereed conference proceedings of the 12th International Conference on Multi-disciplinary Trends in Artificial Intelligence, MIWAI 2018, held in Hanoi, Vietnam, in November 2018. The 16 full papers presented together with 9 short papers were

carefully reviewed and selected from 65 submissions. They are organized in the following topical sections: control, planning and scheduling, pattern recognition, knowledge mining, software applications, strategy games and others.

2. Record Nr.	UNIORUON00190827
Titolo	Between heaven and hell : the myth of Siberia in Russian culture / Edited by Galya Diment and Yuri Slezkine
Pubbl/distr/stampa	New York, : St. Martin's Press, 1993. X, 278 p. ; 21 cm.
ISBN	03-12-06072-6
Disciplina	957
Soggetti	SIBERIA - Letteratura - Storia
Lingua di pubblicazione	Inglese
Formato	Materiale a stampa
Livello bibliografico	Monografia

3. Record Nr.	UNINA9910971850203321
Autore	Farhang-Boroujeny B
Titolo	Adaptive filters : theory and applications // Behrouz Farhang-Boroujeny
Pubbl/distr/stampa	Chichester, West Sussex, U.K., : Wiley, [2013]
ISBN	1-118-59134-8 1-118-59135-6 1-118-59133-X 1-299-46521-8
Edizione	[2nd ed.]
Descrizione fisica	xx, 778 p. : ill
Disciplina	621.3815/324
Soggetti	Adaptive filters Adaptive signal processing
Lingua di pubblicazione	Inglese
Formato	Materiale a stampa
Livello bibliografico	Monografia
Nota di bibliografia	Includes bibliographical references and index.
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Sommario/riassunto

This second edition of Adaptive Filters: Theory and Applications has been updated throughout to reflect the latest developments in this field; notably an increased coverage given to the practical applications of the theory to illustrate the much broader range of adaptive filters applications developed in recent years. The book offers an easy to understand approach to the theory and application of adaptive filters by clearly illustrating how the theory explained in the early chapters of the book is modified for the various applications discussed in detail in later chapters. This integrated approach makes the book a valuable resource for graduate students; and the inclusion of more advanced applications including antenna arrays and wireless communications makes it a suitable technical reference for engineers, practitioners and researchers. Key features: Offers a thorough treatment of the theory of adaptive signal processing; incorporating new material on transform domain, frequency domain, subband adaptive filters, acoustic echo cancellation and active noise control. Provides an in-depth study of applications which now includes extensive coverage of OFDM, MIMO and smart antennas. Contains exercises and computer simulation problems at the end of each chapter. Includes a new companion website hosting MATLAB® simulation programs which complement the theoretical analyses, enabling the reader to gain an in-depth understanding of the behaviours and properties of the various adaptive algorithms.
