

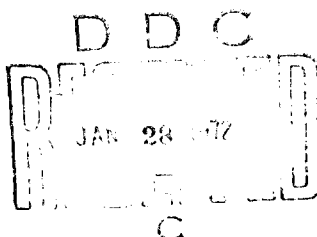
AD735919



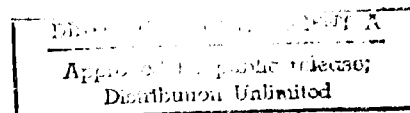
TM-4857/001/00

A SURVEY OF DIGITAL SIGNAL PROCESSING
TECHNIQUES FOR SPEECH ANALYSIS:
INTRODUCTION FOR DOCUMENT SERIES TM-4857

5 January 1972



Reproduced by
NATIONAL TECHNICAL
INFORMATION SERVICE
Springfield, Va 22151



6

Best Available Copy

UNCLASSIFIED

Security Classification

DOCUMENT CONTROL DATA - R & D		
<i>(Security classification of title, body of abstract and indexing annotation must be entered when the overall report is classified)</i>		
1. ORIGINATING ACTIVITY (Corporate author) System Development Corporation Santa Monica, California		2a. REPORT SECURITY CLASSIFICATION Unclassified
		2b. GROUP
3. REPORT TITLE A Survey of Digital Signal Processing Techniques for Speech Analysis: INTRODUCTION for Document Series TM-4857.		
4. DESCRIPTIVE NOTES (Type of report and inclusive dates) Technical Report - July, 1971 to December 1971		
5. AUTHOR(S) (First name, middle initial, last name) H. Barry Ritea		
6. REPORT DATE 5 January 1972	7a. TOTAL NO. OF PAGES 1	7b. NO. OF REFS 0
8a. CONTRACT OR GRANT NO. DAH15-67-C-0149	9a. ORIGINATOR'S REPORT NUMBER(S) TM-4857/001/00	
b. PROJECT NO. ARPA Order No. 1327, Amendment No. 4, c. Program Code No. 2D30 and 2P10.	9b. OTHER REPORT NO(S) (Any other numbers that may be assigned this report)	
d.		
10. DISTRIBUTION STATEMENT Distribution of this document is unlimited.		
11. SUPPLEMENTARY NOTES	12. SPONSORING MILITARY ACTIVITY	
13. ABSTRACT This document provides an introduction for document series TM-4857, A Survey of Digital Signal Processing Techniques for Speech Analysis.		

DD FORM 1 NOV 68 1473

UNCLASSIFIED
Security Classification

14 KEY WORDS	LINK A		LINK B		LINK C	
	ROLE	WT	ROLE	WT	ROLE	WT
Digital Signal Processing Speech Analysis						

TECHNICAL MEMORANDUM

(TM Series)

The work reported herein was supported by the Advanced Research Projects Agency of the Department of Defense under Contract DAHC15-67-C-0149, ARPA Order No. 1327, Amendment No. 4, Program Code No. 2D30 and 2P10.

A SURVEY OF DIGITAL SIGNAL PROCESSING
TECHNIQUES FOR SPEECH ANALYSIS:
INTRODUCTION FOR DOCUMENT SERIES TM-4857

by

H. Barry Ritea

5 January 1972

SYSTEM
DEVELOPMENT
CORPORATION
2500 COLORADO AVE.
SANTA MONICA
CALIFORNIA
90406

The views and conclusions contained in this document are those of the authors and should not be interpreted as necessarily representing the official policies, either expressed or implied of the Advanced Research Projects Agency or the U. S. Government.



Best Available Copy

Distribution of this document is unlimited.

5 January

i
(Page ii blank)

System Development Corporation
TM-4857/001/00

ABSTRACT

This document provides an introduction for document series TM-4857, A Survey of Digital Signal Processing Techniques for Speech Analysis.

1. INTRODUCTION

The analysis of speech characteristically involves the use of digital signal-processing techniques. Although analog methods are preferred for their speed, digital techniques can be modified more easily and can be quickly added to other software packages. Moreover, the computational efficiency of some digital techniques has increased so substantially over the past several years that these techniques have become more and more attractive to the researcher. In particular, the rediscovery in 1965 of the so-called Fast Fourier Transform (FFT) algorithm made the practical computation of the discrete Fourier Transform (DFT) a reality. Using the FFT algorithm and various properties of the DFT, it was later shown that autocorrelation functions, convolutions, and digital filters could be calculated efficiently. More recent interesting applications of the FFT have been directed to the problems of pitch detection and formant analysis of voiced speech.

In this document series we shall present different FFT algorithms and contrast them with respect to computation time, accuracy, storage requirements, and other restrictions. In addition, various applications of the FFT will be given, along with sample test cases. Complete FORTRAN codes will accompany the discussions. As new algorithms are developed, they will be tested as above, and the results will be published in future volumes in this series.

The objective is non-tutorial insofar as the description of the algorithms is concerned. Rather, this series will provide a clearinghouse for the algorithms so that each can be similarly tested and evaluated and the best can be chosen objectively.