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A graphic element consisting of a horizontal line with a series of vertical bars of varying heights on the right side, resembling a signal waveform or a spectrum plot. The bars are arranged in a pattern that suggests a signal being processed or analyzed.

Table of Contents

Volume I

SP1: CELP CODING

Analysis by Synthesis Speech Coding with Generalized Pitch Prediction	1
<i>Paul Mermelstein; Yasheng Qian</i>	
A 16, 24, 32 kbit/s Wideband Speech Codec Based on ATCELP	5
<i>Pierre Combescure; Juergen Schnitzler; Kyrill Fischer; Ralf Kirchherr; Claude Lamblin; Alain Le Guyader; Dominique Massaloux; Catherine Quinquis; Joachim Stegmann; Peter Vary</i>	
A 6.1 to 13.3-kb/s Variable Rate CELP Codec (VR-CELP) for AMR Speech Coding	9
<i>Stefan Heinen; Marc Adrat; Oliver Steil; Peter Vary; Wen Xu</i>	
CELP Speech Coding Based on an Adaptive Pulse Position Codebook	13
<i>Tadashi Amada; Kimio Miseki; Masami Akamine</i>	
A Multistage Search of Algebraic CELP Codebooks	17
<i>Miguel A Ramirez; Max Gerken</i>	
A Fast Search Method Of Algebraic Codebook By Reordering Search Sequence	21
<i>Nam Kyu ha</i>	
An 8 kbit/s ACELP Coder with Improved Background Noise Performance	25
<i>Roar Hagen; Erik Ekudden</i>	
On Phase Perception in Speech	29
<i>Harald Pobloth; W. Bastiaan Kleijn</i>	

SP2: LARGE VOCABULARY RECOGNITION

Progress in Broadcast News Transcription at Dragon Systems	33
<i>Steven Wegmann; Puming Zhan; Larry Gillick</i>	
Recent Improvements to IBM's Speech Recognition System for Automatic Transcription of Broadcast News	37
<i>Scott S Chen; Ellen M Eide; Mark Gales; Ramesh A Gopinath; Dimitri Kanevsky; Peder A Olsen</i>	
Recent Experiments in Large Vocabulary Conversational Speech Recognition	41
<i>Jayadev Billa; Thomas Colhurst; Amro El-Jaroudi; Rukmini Iyer; Kristine Ma; S. Matsoukas; Carl Quillen; Fred Richardson; Manhung Siu; George Zavaliagos; Herb Gish</i>	
Large Vocabulary Speech Recognition in French	45
<i>Martine Adda-Decker; Gilles Adda; Jean-Luc S Gauvain; Lori F Lamel</i>	
The Cambridge University Spoken Document Retrieval System	49
<i>Sue E Johnson; Pierre Jourlin; Gareth L Moore; Karen Spärck Jones; Philip C Woodland</i>	
Improvements in Recognition of Conversational Telephone Speech	53
<i>Barbara Peskin; Michael Newman; Don McAllaster; Venkatesh Nagesha; Hywel Richards; Steven Wegmann; Melvyn Hunt; Larry Gillick</i>	

The 1998 HTK System for Transcription of Conversational Telephone Speech	57
<i>Thomas Hain; Philip C Woodland; Thomas R Niesler; Edward W.D Whittaker</i>	
Real-Time Telephone-Based Speech Recognition in the Jupiter Domain	61
<i>James R Glass; Timothy J Hazen; I. Lee Hetherington</i>	
 SP3: SPEECH ANALYSIS AND ENHANCEMENT	
Template-Driven Generation of Prosodic Information for Chinese Concatenative Synthesis	65
<i>Chung-Hsien Wu; Jau-Hung Chen</i>	
Speech Enhancement Using Nonlinear Microphone Array with Complementary Beamforming	69
<i>Hiroshi Saruwatari; Shoji Kajita; Kazuya Takeda; Fumitada Itakura</i>	
A Multivariate Speech Activity Detector Based on the Syllable Rate	73
<i>David C Smith; Jeffrey Townsend; Douglas J Nelson; Dan Richman</i>	
Discriminating Speakers with Vocal Nodules Using Aerodynamic and Acoustic Features	77
<i>Jeff Kuo; Eva B. Holmberg; Robert E. Hillman</i>	
Enhancement of Esophageal Speech Using Formant Synthesis	81
<i>Kenji Matsui; Noriyo Hara</i>	
Development of Rules for Controlling the HLsyn Speech Synthesizer	85
<i>Helen M Hanson; Richard S McGowan; Kenneth N Stevens; Robert E Beaudoin</i>	
On the Characteristics and Effects of Loudness During Utterance Production in Continuous Speech Recognition	89
<i>Daniel Tapias; Carlos Garcia; Christophe Cazassus</i>	
A Multi-Channel Speech/Silence Detector Based on Time Delay Estimation and Fuzzy Classification	93
<i>Francesco Beritelli; Salvatore Casale; Alfredo Cavallaro</i>	
Noise Suppression Using A Time-Varying, Analysis/Synthesis Gammachirp Filterbank	97
<i>Toshio Irino</i>	
Experimental Comparison of Signal Subspace Based Noise Reduction Methods	101
<i>Peter S. K. Hansen; Per C. Hansen; Steffen D. Hansen; John A Sørensen</i>	
 SP4: ACOUSTIC MODELING I	
Using a Large Vocabulary Continuous Speech Recognizer for a Constrained Domain with Limited Training	105
<i>Man-hung Siu; Michael Jonas; Herbert Gish</i>	
Initial Evaluation of Hidden Dynamic Models on Conversational Speech	109
<i>Joseph Picone; Sandi Pike; Roland Reagan; Terri Kamm; John Bridle; Li Deng; Z. Ma; Hywel Richards; Mike Schuster</i>	
Convolutional Density Estimation in Hidden Markov Models for Speech Recognition	113
<i>Spyros Matsoukas; George Zavaliagos</i>	
Automatic Clustering and Generation of Contextual Questions for Tied States in Hidden Markov Models	117
<i>Rita Singh; Bhiksha Raj; Richard M Stern</i>	

Partly Hidden Markov Model and its Application to Speech Recognition	121
<i>Tetsunori Kobayashi; Junko Furuyama; Ken Masumitsu</i>	
Hidden Markov Models with Divergence Based Vector Quantized Variances	125
<i>Jae H Kim; Raziel Haimi-Cohen; Frank K Soong</i>	
HMM Training Based on Quality Measurement	129
<i>Yuqing Gao; Ea-Ee Jan; Mukund Padmanabhan; Michael Picheny</i>	
Prosodic Word Boundary Detection Using Statistical Modeling of Moraic Fundamental Frequency Contours and Its Use for Continuous Speech Recognition	133
<i>Koji Iwano; Keikichi Hirose</i>	
 SP5: ASR SYSTEMS AND APPLICATIONS	
Voice Recognition Focusing on Vowel Strings on a Fixed-Point 20-Mips DSP Board	137
<i>Yukikuni Nishida; Yoshio Nakadai; Yoshitake Suzuki; Toshihide Kurokawa; Hirokazu Sato; Tetsuma Sakurai</i>	
Speech Interface VLSI for Car Applications	141
<i>Makoto Shozakai</i>	
Recognition of Elderly Speech and Voice-Driven Document Retrieval	145
<i>Stephen W Anderson; Natalie Liberman; Erica Bernstein; Stephen Foster; Erin Cate; Brenda Levin; Randy Hudson</i>	
A Comparison of Features for Speech, Music Discrimination	149
<i>Michael J Carey; Eluned S Parris; Harvey Lloyd-Thomas</i>	
Recognizing Connected Digits in a Natural Spoken Dialog	153
<i>Mazin G Rahim</i>	
Telephone Speech Recognition Using Neural Networks and Hidden Markov Models	157
<i>DongSuk Yuk; James L Flanagan</i>	
Improving Speech Recognition Performance By Using Multi-Model Approaches	161
<i>Ji Ming; Philip Hanna; Darryl Stewart; Marie Ownes; F. Jack Smith</i>	
Speaker-Dependent Name Dialing in a Car Environment with Out-of-Vocabulary Rejection	165
<i>C. S Ramalingam; Yifan Gong; Lorin P Netsch; Wallace W Anderson; John J Godfrey; Yu-Hung Kao</i>	
A New Method Used in HMM for Modeling Frame Correlation	169
<i>Qing Guo; Fang Zheng; Jian Wu; Wenhui Wu</i>	
N-Best Based Supervised and Unsupervised Adaptation for Native and Non-Native Speakers in Cars	173
<i>Patrick Nguyen; Philippe Gelin; Jean-Claude Junqua; Jen-Tzung Chien</i>	
 SP6: TOPICS IN SPEECH CODING	
Performance Assessment of Tandem Connection of Enhanced Cellular Coders	177
<i>Simao F Campos Neto; Franklin L Corcoran; Ara Karahisar</i>	
TTS Based Very Low Bit Rate Speech Coder	181
<i>Ki-Seung Lee; Richard V. Cox</i>	
Wideband Speech Coding With Toll Quality Based on IA-Model	185
<i>Ling Kok Ng; Gang Li; Xiao Lin; Guoan Bi</i>	

4 kb/s Multi-Pulse Based CELP Speech Coding Using Excitation Switching	189
<i>Kazunori Ozawa</i>	
An Adaptive Multi-Rate Speech Coder For Digital Cellular Telephony	193
<i>Erdal Paksoy; Juan Carlos De Martin; Alan V McCree; Christian G Gerlach;</i> <i>Anand Anandakumar; Wai-Ming Lai; Vishu Viswanathan</i>	
An Adaptive Post-Filtering Technique Based on The Modified Yule-Walker Filter	197
<i>Azhar Mustapha; Suat Yeldener</i>	
A Modular Approach to Speech Enhancement with an Application to Speech Coding	201
<i>Anthony J Accardi; Richard V Cox</i>	
On Speech Coding in a Perceptual Domain	205
<i>Gernot Kubin; W. Bastiaan Kleijn</i>	
 SP7: SPEECH ANALYSIS	
Speech Analysis/Synthesis/Conversion by Using Sequential Processing	209
<i>Panuthat Boonpramuk; Tetsuo Funada; Noboru Kanedera</i>	
Modelling Energy Flow in the Vocal Tract with Applications to Glottal Closure and Opening Detection	213
<i>Mike Brookes; Han Pin Loke</i>	
Fitting the Mel Scale	217
<i>Srinivasan Umesh; Leon Cohen; Douglas Nelson</i>	
Fast Accent Identification and Accented Speech Recognition	221
<i>Pascale Fung; Wai Kat LIU</i>	
Relevancy of Time-Frequency Features for Phonetic Classification Measured by Mutual Information	225
<i>Howard H Yang; Sarel J Van Vuuren; Hynek Hermansky</i>	
Hidden Markov Models Based on Multi-Space Probability Distribution for Pitch Pattern Modeling	229
<i>Keiichi Tokuda; Takashi Masuko; Noboru Miyazaki; Takao Kobayashi</i>	
An Algorithm for Glottal Volume Velocity Estimation	233
<i>Ashraf Alkhairy</i>	
Frame-Level Noise Classification in Mobile Environments	237
<i>Khaled El-Maleh; Ara Samouelian; Peter Kabal</i>	
 SP8: LOW BIT RATE SPEECH CODING I	
Low Delay Multi-level Decomposition and Quantisation Techniques for WI Coding	241
<i>Nicola R Chong; Ian S Burnett; Joe F Chicharo</i>	
An Improved Mixed Excitation Linear Prediction (MELP) Coder	245
<i>Takahiro Unno; Thomas P Barnwell III; Kwan Truong</i>	
Split Band LPC Based Adaptive Multi-Rate GSM Candidate	249
<i>Stephane Villette; Milos Stefanovic; Ahmet Kondo</i>	
Frequency-Domain Spectral Envelope Estimation for Low Rate Coding of Speech	253
<i>Milan Jelinek; Jean-Pierre Adoul</i>	

Robust Closed-Loop Pitch Estimation for Harmonic Coders by Time Scale Modification	257
<i>Chunyan Li; Vladimir Cuperman; Allen Gersho</i>	
Phase Adjustment in Waveform Interpolation	261
<i>Hong-Goo Kang; D. Sen</i>	
A Low Resolution Pulse Position Coding Method for Improved Excitation Modeling of Speech Transition	265
<i>Jongseo Sohn; Wonyong Sung</i>	
Dispersion Phase Vector Quantization For Enhancement of Waveform Interpolative Coder	269
<i>Oded Gottesman</i>	
 SP9: ROBUST SPEECH RECOGNITION IN NOISY ENVIRONMENTS	
The Teager Energy Based Feature Parameters for Robust Speech Recognition in Car Noise	273
<i>Firas Jabloun; Enis A Cetin</i>	
Avoiding Distortions Due to Speech Coding and Transmission Errors in GSM ASR Tasks	277
<i>Ascensi—n Gallardo-Antol'n; Fernando D'az-de-Mar'a; Francisco Valverde-Albacete</i>	
Binaural Bark subband Preprocessing of Nonstationary Signals for Noise Robust Speech Feature Extraction	281
<i>Mike Peters</i>	
Speaker Normalized Spectral Subband Parameters for Noise Robust Speech Recognition	285
<i>Satoru Tsuge; Toshiaki Fukada; Harald Singer</i>	
Temporal Patterns (TRAPs) In ASR of Noisy Speech	289
<i>Hynek Hermansky; Sangita Sharma</i>	
Signal Modeling for Isolated Word Recognition	293
<i>Montri Karnjanadecha; Stephen A Zahorian</i>	
Transforming HMMs For Speaker-Independent Hands-Free Speech Recognition in the Car	297
<i>Y. Gong; John J. Godfrey</i>	
Channel and Noise Adaptation via HMM Mixture Mean Transform and Stochastic Matching	301
<i>Shuen Kong Wong; Bertram Shi</i>	
 SP10: SPEAKER RECOGNITION	
Speaker Verification Performance and the Length of Test Sentence	305
<i>Jialong He; Li Liu</i>	
On The Use of Some Divergence Measures in Speaker Recognition	309
<i>Rivarol Vergin; Douglas O'Shaughnessy</i>	
Improving a GMM Speaker Verification System by Phonetic Weighting	313
<i>Roland Auckenthaler; Eluned S Parris; Michael J Carey</i>	
A Hybrid Score Measurement For HMM-Based Speaker Verification	317
<i>Yong Gu; Trevor Thomas</i>	
Polynomial Classifier Techniques for Speaker Verification	321
<i>William M Campbell; Khaled T Assaleh</i>	

Channel-Robust Speaker Identification Using Modified-Mean Cepstral Mean Normalization with Frequency Warping	325
<i>Alvin A Garcia; Richard J Mammon</i>	
Feature Selection Using Genetics-Based Algorithm and Its Application to Speaker Identification	329
<i>Mubecce Demirekler; Ali Haydar</i>	
 SP11: ACOUSTIC MODELING II	
Frame Discrimination Training of HMMs for Large Vocabulary Speech Recognition	333
<i>Dan Povey; Philip C Woodland</i>	
Discriminative Mixture Weight Estimation for Large Gaussian Mixture Models	337
<i>Francoise Beaufays; Mitchel Weintraub; Yochai Konig</i>	
Modeling Disfluency and Background Events in ASR for a Natural Language Understanding Task	341
<i>Richard C. Rose; Giuseppe Riccardi</i>	
Decision Tree State Tying Based on Penalized Bayesian Information Criterion	345
<i>Wu Chou; Wolfgang Reichl</i>	
A 2D Extended HMM for Speech Recognition	349
<i>Jiayu Li; Alejandro Murua</i>	
Probabilistic Classification of HMM States for Large Vocabulary Continuous Speech Recognition	353
<i>Xiaoqiang Luo; Frederick Jelinek</i>	
The HDM: A Segmental Hidden Dynamic Model of Coarticulation	357
<i>Hywel B Richards; John S Bridle</i>	
Maximum Likelihood Estimates for Exponential Type Density Families	361
<i>Sankar Basu; Charles A Micchelli; Peder A Olsen</i>	
 SP12: SPEECH PRODUCTION AND SYNTHESIS	
On the Limits of Speech Recognition in Noise	365
<i>Stephen D Peters; Peter Stubbley; Jean-Marc Valin</i>	
Recognition of Spectrally Degraded Speech in Noise with Nonlinear Amplitude Mapping	369
<i>Qian-Jie Fu; Robert V. Shannon</i>	
Phrase Splicing and Variable Substitution Using the IBM Trainable Speech Synthesis System	373
<i>Robert E Donovan; Martin Franz; Jeffrey S Sorensen; Salim Roukos</i>	
Assessment and Correction of Voice Quality Variabilities in Large Speech Databases for Concatenative Speech Synthesis	377
<i>Yannis G Stylianou</i>	
Shape Invariant Time-Scale Modification of Speech Using a Harmonic Model	381
<i>Darragh O'Brien; Alex Monaghan</i>	
Using a Sigmoid Transformation for Improved Modeling of Phoneme Duration	385
<i>Kim E.A Silverman; Jerome R Bellegarda</i>	
Nonlinear Dynamic Modeling of the Voiced Excitation for Improved Speech Synthesis	389
<i>Karthik Narasimhan; Jose C. Principe; Donald G. Childers</i>	

Results on Perceptual Invariants to Transformations on Speech	393
<i>Arnaud Robert</i>	

SP13: FEATURE EXTRACTION

Investigations on Inter-Speaker Variability in the Feature Space	397
<i>Reinhold Haeb-Umbach</i>	

LSP Weighting Functions Based on Spectral Sensitivity and Mel-Frequency Warping for Speech Recognition in Digital Communication	401
<i>Seung Ho Choi; Hong Kook Kim; Hwang Soo Lee</i>	

Two-Dimensional Multi-Resolution Analysis of Speech Signals and its Application to Speech Recognition	405
<i>Chun-ping Chan; Yiu-wing Wong; Tan Lee; Pak-chung Ching</i>	

Hierarchical Subband Linear Predictive Cepstral (HSLPC) Features for HMM-Based Speech Recognition	409
<i>Rathinavelu Chengalvarayan</i>	

Towards a Robust/Fast Continuous Speech Recognition System Using a Voiced-Unvoiced Decision	413
<i>Douglas O'Shaughnessy; Hesham Tolba</i>	

A C/V Segmentation Algorithm For Mandarin Speech Signal Based on Wavelet Transforms	417
<i>Jhing-Fa Wang; Shi-Huang Chen</i>	

Feature Extraction for Speech Recognition Based on Orthogonal Acoustic Feature Planes and LDA	421
<i>Tsuneo Nitta</i>	

Distinctive Feature Detection Using Support Vector Machines	425
<i>Partha Niyogi; Chris Burges; Padma Ramesh</i>	

SP14: ROBUST SPEECH RECOGNITION AND ADAPTATION

Time-Varying Noise Compensation Using Multiple Kalman Filters	429
<i>Nam Soo Kim</i>	

A Segment-Based C0 Adaptation Scheme for PMC-based Noisy Mandarin Speech Recognition	433
<i>Wei-Tyng Hong; Sin-Horng Chen</i>	

Improved Parallel Model Combination Techniques With Split Gaussian Mixtures For Speech Recognition Under Noisy Conditions	437
<i>Jeih-weih Hung; Jia-lin Shen; Lin-shan Lee</i>	

Speech Recognition and Enhancement by A Nonstationary AR HMM with Gain Adaptation under Unknown Noise	441
<i>Ki Yong Lee; Joohun Lee; Gunther Ruske</i>	

Database and Online Adaptation for improved Speech Recognition in Car Environments	445
<i>Alexander Fischer; Volker Stahl</i>	

Training of HMM with Filtered Speech Material for Hands-free Recognition	449
<i>Diego Giuliani; Marco Matassoni; Maurizio Omologo; Piergiorgio Svaizer</i>	

Incremental Enrollment of Speech Recognizers	453
<i>Chafic E Mokbel; Olivier Collin</i>	

Automatic Speech Recognition: a Communication Perspective	457
<i>Bishnu S Atal</i>	

SP15: LOW BIT RATE SPEECH CODING II

Split Band CELP (SB-CELP) Speech Coder	461
<i>Mohammad R Nakhai; Farokh A Marvasti</i>	
Log Amplitude Modeling of Sinusoids in Voiced Speech	465
<i>Najam Malik; W. Harvey Holmes</i>	
1.2kbit/s Harmonic Coder Using Auditory Filters	469
<i>Minoru Kohata</i>	
Exponential Sinusoidal Modeling of Transitional Speech Segments	473
<i>Jesper Jensen; Søren H Jensen; Egon Hansen</i>	
Harmonic+Noise Coding Using Improved V/UV Mixing and Efficient Spectral Quantization	477
<i>Eric W. M. Yu; Cheung-Fat Chan</i>	
A 4 Kb/s Toll Quality Harmonic Excitation Linear Predictive Speech Coder	481
<i>Suat Yeldener</i>	
High Quality MELP Coding at Bit-Rates Around 4 kb/s	485
<i>Jacek Stachurski; Alan McCree; Vishu Viswanathan</i>	
Pitch Quantization in Low Bit-Rate Speech Coding	489
<i>Thomas Eriksson; Hong-Goo Kang</i>	

SP16: SPEECH UNDERSTANDING

Incorporating Confidence Measures in the Dutch Train Timetable Information System Developed in the Arise Project	493
<i>Gies Bouwman; Janienke Sturm; Louis Boves</i>	
HMM and Neural Network Based Speech Act Detection	497
<i>Klaus Ries</i>	
The LIMSI ARISE System for Train Travel Information	501
<i>Lori F Lamel; Sophie Rosset; Jean-Luc S Gauvain; Samir K Bennacef</i>	
Improving The Suitability Of Imperfect Transcriptions For Information Retrieval From Spoken Documents	505
<i>Matthew A Siegler; Michael J. Witbrock</i>	
Automatic Topic Identification for Two-Level Call Routing	509
<i>John A Golden; Owen Kimball; Man-Hung Siu; Herbert Gish</i>	
Named Entity Tagged Language Models	513
<i>Yoshihiko Gotoh; Steve Renals; Gethin Williams</i>	
Speech Translation: Coupling of Recognition and Translation	517
<i>Hermann Ney</i>	
Probabilistic Models for Topic Detection and Tracking	521
<i>Frederick G Walls; Hubert Jin; Sreenivasa Sista; Richard Schwartz</i>	

SP17: LANGUAGE MODELING I

Discriminative Estimation of Interpolation Parameters for Language Model Classifiers	525
<i>Volker Warnke; Stefan Harbeck; Elmar Noeth; Heinrich Niemann; Michael Levit</i>	
Combination of Words and Word Categories in Varigram Histories	529
<i>Reinhard Blasig</i>	
Multi-Class Composite N-gram Based on Connection Direction	533
<i>Hirofumi Yamamoto; Yoshinori Sagisaka</i>	
A Class-Based Language Model for Large-Vocabulary Speech Recognition Extracted from Part-of-Speech Statistics	537
<i>Christer Samuelsson; Wolfgang Reichl</i>	
Improved Topic-Dependent Language Modeling Using Information Retrieval Techniques	541
<i>Milind Mahajan; Doug Beeferman; X.D. Huang</i>	
Smoothing Methods in Maximum Entropy Language Modeling	545
<i>Sven C Martin; Hermann Ney; Joerg Zaphlo</i>	
Efficient Sampling and Feature Selection in Whole Sentence Maximum Entropy Language Models	549
<i>Stanley Chen; Ronald Rosenfeld</i>	
A Maximum Entropy Language Model Integrating N-gram and Topic Dependencies for Conversational Speech Recognition	553
<i>Sanjeev P Khudanpur; Jun Wu</i>	

Volume II

SP18: ACOUSTIC MODELING III

Connected Digit Recognition Using Short and Long Duration Models	557
<i>Cristina Chesta; Pietro Laface; Franco Ravera</i>	
Discriminative Training Via Linear Programming	561
<i>Kishore A Papineni</i>	
Refining Tree-Based Clustering by Means of Formal Concept Analysis, Balanced Decision Trees and Automatically Generated Model-Sets	565
<i>Daniel Willett; Christoph Neukirchen; Jörg Rottland; Gerhard Rigoll</i>	
Efficient Speech Recognition Using Subvector Quantization And Discrete-Mixture HMMs	569
<i>Stavros Tsakalidis; Vassilios Digalakis; Leonardo G Neumeyer</i>	
A Unified Approach of Incorporating General Features in Decision Tree Based Acoustic Modeling	573
<i>Wolfgang Reichl; Wu Chou</i>	
Irrelevant Variability Normalization in Learning HMM State Tying From Data Based on Phonetic Decision-Tree	577
<i>Qiang Huo; Bin Ma</i>	
Discriminative Spectral-Temporal Multi-Resolution Features for Speech Recognition	581
<i>Philip McMahon; Naomi Harte; Saeed Vaseghi; Paul McCourt</i>	