

Contents

Acknowledgments	i
Abstract	iii
Kurzfassung	v
1 Introduction	1
1.1 Concept of Robust Source Decoding	1
1.2 Brief Overview of Some Important Speech and Audio Codecs	2
1.3 Outline of the Thesis	3
2 Aspects of Source Parameter Encoding and Decoding	5
2.1 Source Parameter Models	5
2.1.1 Modeling Unquantized Parameters	5
2.1.2 Quantization of Parameters	6
2.1.3 Modeling Quantized Parameters	7
2.2 Channel Models	9
2.2.1 Additive White Gaussian Noise (AWGN) Channel With Binary Phase-Shift Keying (BPSK)	10
AWGN Channel	10
BPSK Modulation	10
2.2.2 Gilbert-Elliott Channel	11
2.3 Fixed-Length (FL) Soft-Decision (SD) Decoding of Parameters	12
2.3.1 Hard-Decision (HD) Decoding	13
2.3.2 SD: <i>A Posteriori</i> Probabilities (APPs)	13
Channel Transition Probabilities	15
Predictive <i>A Priori</i> Probabilities	15
<i>A Posteriori</i> Probabilities (APPs)	16
Forward-Backward Algorithm (BCJR Algorithm)	16
2.3.3 SD: Parameter Estimation	17
2.4 Summary	17
3 Improving AMR Narrowband and Wideband by Fixed-Length Soft- Decision Decoding	19
3.1 AMR Speech Coding	19
3.1.1 AMR Narrowband	20

	AMR-NB Encoding	20
	AMR-NB Decoding	20
	Line Spectral Frequencies (LSFs)	21
	Adaptive Codebook Index (Pitch Delay)	21
	Adaptive Codebook and Fixed Codebook Gains	22
	Fixed Codebook Index	22
	Bit Allocation Table	22
3.1.2	AMR Wideband	22
	AMR-WB Encoding	23
	AMR-WB Decoding	23
	Immittance Spectral Frequencies (ISFs)	23
	Adaptive Codebook Index (Pitch Delay)	24
	Vector-Quantized Codebook Gain (VQ Gain)	24
	Fixed Codebook Index	24
	Bit Allocation Table	24
3.2	Standard Error Concealment in AMR	25
3.2.1	AMR Narrowband	25
3.2.2	AMR Wideband	25
3.3	New AMR Parameter Soft-Decision Decoding	25
3.3.1	AMR Narrowband	26
	Line Spectral Frequencies (LSFs)	27
	Adaptive Codebook Index (Pitch Delay)	29
	Adaptive Codebook and Fixed Codebook Gains	30
	Complexity Evaluation	30
3.3.2	AMR Wideband	31
	Immittance Spectral Frequencies (ISFs)	32
	Adaptive Codebook Index (Pitch Delay)	33
	Vector-Quantized Codebook Gain (VQ Gain)	33
	Other Parameters	34
	Complexity Evaluation	34
3.4	Simulation Setup and Results	34
3.4.1	Simulation Setup	35
3.4.2	Simulation Results	35
	AMR Narrowband	36
	AMR Wideband	40
3.5	Summary	43
4	Variable-Length Soft-Decision Decoding	45
4.1	Introduction	45
4.2	Variable-Length (VL) Hard-Decision Decoding	48
4.3	Soft-Decision (SD) Decoding: Trellis Representation	49
	Definitions of Stage Boundaries and State Intervals	50
4.4	SD: <i>A Posteriori</i> Probabilities and Source Symbol Estimation	53
4.4.1	<i>A Posteriori</i> Probabilities (APPs)	53
	Forward Recursion	53

Backward Recursion	55
4.4.2 Source Symbol Estimation	56
4.5 From VL/SD Decoding to FL/SD Decoding	56
APPs With AK0	57
APPs With AK1	57
4.6 Simulation Setup and Results	58
4.6.1 Simulation Setup	58
4.6.2 Simulation Results	58
4.7 Summary	62
5 Improving HE-AAC by Soft-Decision Decoding	65
5.1 Introduction	65
5.2 HE-AAC Audio Coding	67
5.2.1 AAC Encoding	67
Filterbank	67
Psychoacoustic Model and Reduction of Psychoacoustic Requirements	68
Determination of Scale Factors and Quantization of Spectral Coef-	
ficients	68
Noiseless Coding	73
Out of Bits Prevention	76
5.2.2 AAC Decoding	76
5.2.3 HE-AAC Bit Stream Structure	77
5.3 Standard Error Concealment in HE-AAC	79
5.4 New HE-AAC Parameter Soft-Decision (SD) Decoding	79
5.4.1 FL/SD Decoding of the Global Gain	80
5.4.2 VL/SD Decoding of the Scale Factors	80
5.4.3 VL/SD Decoding of the Quantized Spectral Coefficients	81
New Trellis Representation	81
<i>A Posteriori</i> Probabilities and Source Symbol Estimation	84
5.5 Simulation Setup and Results	86
5.5.1 Simulation Setup	86
Instrumental Measurement and Subjective Listening Test	86
5.5.2 Simulation Results	89
Single Parameter Distorted	89
Two Parameters Distorted	92
Three Parameters Distorted	94
Subjective Listening Test Results	96
5.6 Summary	97
6 Low-Rate Fixed-Length Hard- and Soft-Decision Decoding With Time-	99
Variant Codebooks	99
6.1 Introduction	100
6.2 New Predictive Hard-Decision Decoding	102
6.3 Analysis and Optimization of the Prediction Error PDF	104
6.4 New Predictive Soft-Decision Decoding	106

6.5	Simulation Setup and Results	107
6.5.1	Simulation Setup	107
6.5.2	Simulation Results	107
6.6	Summary	112
7	Improving G.726 and G.722 ADPCM by Hard-Decision Decoding With Time-Variant Codebooks	113
7.1	G.726 and G.722 ADPCM Speech Coding	113
7.1.1	G.726	114
	G.726 ADPCM Encoding	114
	G.726 ADPCM Decoding	115
7.1.2	G.722	115
	G.722 ADPCM Encoding	115
	G.722 ADPCM Decoding	116
7.2	New Hard-Decision Decoding With Time-Variant Codebooks	118
7.2.1	Error-Free Transmission Conditions	119
	An Outline	119
	Application to ADPCM	119
	Normalized Least-Mean-Squares (NLMS) Algorithm	121
7.2.2	Hard-Decision Decoding	121
7.3	Simulation Setup and Results	121
7.3.1	Simulation Setup	122
7.3.2	Simulation Results	123
	G.726	123
	G.722	128
7.4	Summary	130
8	Conclusions	131
A	APP Calculation of Exploiting Both Interframe and Intraframe Redundancy in AMR-NB	133
B	Time-Variant Quantization Codebook	137
B.1	New Reconstruction Levels	137
B.2	Receiver-Sided Prediction Error Variance	138
C	Audio Database	141
D	PEAQ ODG Results	143
E	Simulation Results in Tabular Forms	145
	List of Symbols	163
	List of Abbreviations	169
	Bibliography	171

