

Digital Signal Processing

Using MATLAB®

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Contents

List of Abbreviations	xiii
Useful MATLAB Functions	xv
1 Signals	1
1.1 Signals as Waveforms	1
1.2 Applications of Digital Signal Processing	2
1.3 Discrete Versus Continuous Signals	3
1.4 Analog-to-Digital and Digital-to-Analog Conversion	4
1.4.1 Analog-to-Signal Conversion (Signal Digitization)	5
1.4.2 Error Analysis of Analog Signal Digitization	9
1.5 Time Shifting a Signal	11
1.6 Special Functions	12
1.6.1 Sinusoidal Signal	12
1.6.2 Relating the Phase to the Time Shift	13
1.6.3 Unit Step (Heaviside) Function	15
1.6.4 Unit Pulse Function	16
1.6.5 Dirac Delta Function	17
1.7 Problems	17
2 Sinusoidal Signals	21

2.1	Cosine Function	21
2.2	Sinusoidal Signal	22
2.2.1	Relating the Phase to Time Shift	23
2.3	Euler's Formula & Inverse Euler's Formula	25
2.4	Expressing A Sinusoid as Real Part of a Complex Exponential	26
2.5	Phasor Representation of a Sinusoidal Signal	27
2.6	Two-Sided Spectrum of a Sinusoid: Inverse Euler's Formula	28
2.7	More Complex Signals	31
2.7.1	Product of Two Sinusoidal Signals: Amplitude Modulation	32
2.7.2	Superheterodyne Receivers/Mixer	33
2.7.3	Sine Squared Signal	33
2.8	Problems	34
3	Sampling and Aliasing	39
3.1	Sampling of Analog Signals	39
3.2	Sampling a Sinusoidal Signal: Normalized Radian Frequency	41
3.3	Aliasing	44
3.4	Range of Normalized Radian Frequency	45
3.4.1	Spectrum of a Discrete-Time Sinusoid	46
3.5	The Sampling Theorem (Nyquist Rate)	47
3.6	Converting Continuous-Time to Discrete-Time Signals and Vice Versa	48
3.7	Effects of the Sampling Rate	48
3.7.1	Case when $0 \leq \hat{\omega}_0 < \pi$: Oversampling	48
3.7.2	Case when $\pi \leq \hat{\omega} < 2\pi$: Undersampling with Folding	49
3.8	Case when $\hat{\omega} \geq 2\pi$: Undersampling	51
3.9	Problems	52
4	Discrete-Time Systems	57

4.1	Special Signals	57
4.1.1	Unit Impulse or Kronecker Delta $\delta[n]$	57
4.1.2	Unit Step $u[n]$	58
4.1.3	Length L Rectangular Pulse $r_L[n]$	59
4.1.4	Real Exponential Signal	61
4.1.5	Complex Exponential Signal	61
4.1.6	Linear Frequency-Modulated (LFM) Chirp Signal	62
4.2	Signal Energy	63
4.3	Signal Power	64
4.4	Power of a Periodic Signal	65
4.4.1	Sampling Rate Restriction for Periodic Signals	67
4.5	System Example: Digital Moving Average (MA) Filter	69
4.6	Causal and Noncausal Systems	71
4.7	Linear and Nonlinear Systems	72
4.8	Shift-Invariant and Shift-Variant Systems	73
4.9	Nonrecursive and Recursive Systems	76
4.10	Expressing a Discrete-Time Signal as a Sum of Scaled and Delayed Impulses	77
4.11	The Impulse Response of Nonrecursive Systems	80
4.12	Impulse Response of Recursive Systems	83
4.13	Convolution Operation	85
4.13.1	Linearity Property of Convolution	85
4.13.2	Commutative Property of Convolution	86
4.13.3	Distributive Property of Convolution	86
4.13.4	Associative Property of Convolution	87
4.13.5	FIR is a Convolution	88
4.13.6	Computing Output of Convolution Using MATLAB	89
4.14	Convolution Operation Dependence Graph	89

4.14.1	Length of the Convolution	90
4.14.2	Extracting Impulse Response Using Convolution	92
4.15	Cascaded LSI Systems	92
4.16	Correlation Operation	95
4.16.1	Correlation Operation Dependence Graph	97
4.16.2	Range of the Correlation Shift $c_{xy}[n]$	98
4.17	Matched Filters	102
4.17.1	Matched Filter Dependence Graph	103
4.17.2	Application of Correlation in Radars	103
4.17.3	MATLAB Implementation of Correlation Operation	104
4.18	Problems	107
5	Discrete-Time Fourier Transform (DTFT)	113
5.1	The Forward Discrete-Time Fourier Transform	114
5.2	DTFT of Some Simple Signals	115
5.2.1	DTFT of Unit Impulse	115
5.2.2	DTFT of Length- L Pulse	116
5.2.3	DTFT of Right-sided Decaying Real Exponential Signal	118
5.2.4	DTFT of Right-sided Decaying Complex Exponential Signal	119
5.3	Mathematical Expressions of a Discrete-Time Signal in the Time and Frequency Domains	120
5.4	Properties of the DTFT	121
5.4.1	Linearity Property of DTFT	121
5.4.2	Time Shift Property of DTFT	122
5.4.3	Periodicity Property of DTFT	124
5.4.4	Frequency Shift Property of DTFT	124
5.4.5	Conjugate Symmetry Property of DTFT for Real Signals	125
5.4.6	Time-Reversal Property of DTFT	126

5.5	Relationship Between DTFT and Convolution	127
5.6	The Inverse DTFT (IDTFT)	128
5.6.1	Inverse DTFT of a Constant in the Frequency Domain	128
5.6.2	Inverse DTFT of a Complex Exponential in the Frequency Domain	129
5.6.3	DTFT of a Constant in the Time Domain	130
5.6.4	DTFT of a Double-Sided Complex Exponential in the Time Domain	131
5.6.5	DTFT of Unit Step Function in the Time Domain	132
5.6.6	The Convolution Theorem: Convolution in the Frequency Domain	134
5.7	DTFT of the FIR Filter (the Frequency Response)	135
5.8	Finding the Output of an FIR Filter Given its Impulse Response	136
5.8.1	Method 1: Evaluating $y[n]$ when Impulse Response $h[n]$ is Given	136
5.8.2	Method 2: Evaluating $y[n]$ when Frequency Response $H(\hat{\omega})$ is Given	138
5.8.3	Method 3: Evaluating $y[n]$ when Input is Double-Sided Sinusoid	139
5.9	DTFT of the IIR Filter (the Frequency Response)	142
5.10	Moving Average Filter Revisited	143
5.11	DTFT of Some Ideal Filters	145
5.11.1	Frequency Response of Ideal Lowpass Filter (LPF)	146
5.11.2	Frequency Response of Ideal Highpass Filter (HPF)	147
5.11.3	Frequency Response of Ideal Bandpass Filter (BPF)	149
5.11.4	Frequency Response of Ideal Bandstop Filter (BSF)	151
5.12	Parseval's Theorem	152
5.13	Practical Implementation of the Ideal Lowpass Filter: Filter Windows	155
5.13.1	Making the Filter Coefficients Finite	156
5.13.2	Making the Filter Causal	158
5.14	Practical Implementation of the Ideal Highpass Filter	163
5.15	Practical Implementation of the Ideal Bandpass Filter	164
5.16	Practical Implementation of the Ideal Bandstop Filter	169

5.17	Gibbs Phenomenon in the Time Domain	173
5.18	Filter Design Using MATLAB	174
5.19	Problems	180
6	Discrete Fourier Transform (DFT)	185
6.1	The Forward DFT	185
6.1.1	The Range of k and $\hat{\omega}_k$	186
6.2	DFT of Some Simple Periodic Signals	188
6.2.1	DFT of Periodic Unit Impulse Signal	188
6.2.2	DFT of Delayed Periodic Unit Impulse Signal	188
6.2.3	DFT of Periodic Length- L Pulse $r_L[n]$	189
6.2.4	DFT of Periodic Length- N Pulse	190
6.2.5	DFT of Periodic Exponential Signal	190
6.2.6	DFT of Periodic Complex Exponential Signal	191
6.2.7	DFT of a Periodic Complex Exponential Sequence	191
6.3	DFT of Periodic Cosine Signal	192
6.4	Relation Between DTFT and DFT	193
6.5	Mathematical Expressions of a Discrete-Time Signal in the Time and Frequency Domains	194
6.6	Properties of the DFT	194
6.6.1	Linearity Property of DFT	194
6.6.2	Shift Property of DFT: Circular Shift	196
6.6.3	Periodicity Property of DFT	198
6.6.4	Frequency Shift Property of DFT	199
6.6.5	Conjugate Symmetry of DFT for Real Signals	199
6.6.6	Time-Reversal Property of DFT	200
6.7	Calculating DFT Using MATLAB	202
6.7.1	Changing the Range of k from $0 \leq k < N$ to $-N/2 \leq k < N/2$	202

6.7.2	Changing the Range of k from $-N/2 \leq k < N/2$ to $0 \leq k < N$	203
6.7.3	Centering The Spectrum Around 0 Value	204
6.8	The Inverse DFT	207
6.8.1	Inverse DFT of Impulse $\delta[n]$	207
6.9	Relationship Between DFT and Convolution: Circular Convolution	208
6.9.1	The Convolution Theorem: Convolution in the Frequency Domain	211
6.9.2	Calculating Circular Convolution Using MATLAB	212
6.10	Parseval's Theorem for the DFT	213
6.11	Problems	214
7	z-Transform	219
7.1	The z -Transform Equation	219
7.2	The Region of Convergence (ROC)	220
7.3	Relating z -Transform to Laurent Series and DTFT	223
7.4	z -Transform of Simple Functions	224
7.4.1	z -Transform of Kronecker Delta Impulse Signal	224
7.4.2	z -Transform of Delayed Kronecker Delta Impulse	224
7.4.3	z -Transform of Unit Step	224
7.4.4	z -Transform of Delayed Unit Step	225
7.4.5	z -Transform of Length- L Pulse	226
7.4.6	z -Transform of Decaying Real Exponential Signal	226
7.4.7	z -Transform of Decaying Complex Exponential Signal	227
7.4.8	z -Transform of Complex Exponential Signal	227
7.5	Properties of the z -Transform	229
7.5.1	Linearity Property of the z -Transform	229
7.5.2	Time Shift Property of the z -Transform	230
7.5.3	Initial Value Theorem of z -Transform	231

7.6	Convolution and the z -Transform	231
7.7	FIR System Function	233
7.8	IIR System Function	234
7.9	Poles and Zeros of the System Function	236
7.10	Relation Between the z -Transform, DTFT and DFT	237
7.11	Inverse z -Transform	239
7.11.1	Inverse z -Transform By Inspection when $H(z) = B(z)$	240
7.11.2	Inverse z -Transform Using Polynomial Division	241
7.11.3	Inverse z -Transform Using Partial Fraction Expansion	243
7.11.4	Case of Simple Poles	244
7.11.5	Partial Fractions for Repeated Poles	248
7.11.6	Inverse z -Transform when $N = M$	252
7.11.7	General Expression for Inverse z -Transform Using Complex Analysis	258
7.12	Relation Between the z -Transform and the Laplace Transform	260
7.13	Finding z -Transform Using MATLAB	261
7.13.1	Plotting the Locations of Poles and Zeros	262
7.14	Transient & Steady State Response of an FIR Filter	263
7.15	Transient, Steady State Response and Stability of an IIR Filter	264
7.15.1	IIR Filter Stability	265
7.15.2	IIR Transient Response	265
7.15.3	IIR Steady State Response	266
7.16	Properties of Region of Convergence (ROC) of the z -Transform	266
7.17	Problems	267
8	Digital Filters	273
8.1	Bode Diagrams for $H(\hat{\omega})$	275
8.1.1	Bode Diagrams Using MATLAB	275

8.2	Phase and Group Delays	276
8.2.1	Extracting Phase & Group Delays from the Frequency Response	277
8.3	Digital Filter Design	278
8.3.1	Design of Finite Impulse Response (FIR) Filter	280
8.3.2	Design of Infinite Impulse Response (IIR) Filter	280
8.4	Moving Average Filter (Resonator Filter)	282
8.5	Resonator Filter at $\hat{\omega}_0 > 0$	285
8.6	Notch Filter (Two-Zero Filter)	286
8.7	Single-Pole Recursive Filter	288
8.8	Approximating a Differentiator Filter	291
8.9	Approximating an Integrator Filter	291
8.10	Comb Filters	292
8.11	Gibbs Phenomenon In the Frequency Domain	294
8.11.1	Comparison Between FIR and IIR Filter	295
8.12	Problems	295
9	The Fast Fourier Transform (FFT) Algorithm	299
9.1	Introduction	299
9.2	Decimation-in-Time (DIT) FFT Algorithm	301
9.3	Decimation-in-Frequency (DIF) FFT Algorithm	304
9.4	MATLAB Implementation of the FFT Algorithm	306
9.5	Problems	311
10	Discrete-Time Systems Analysis	313
10.1	Block Diagram and Signal Flow Graph Descriptions	313
10.1.1	FIR Filter Structures	315
10.1.2	IIR Filter Signal Flow Graph Representation	316
10.1.3	Transposed Form for a System Using SFG	318

10.2 State-Space Analysis of Discrete-Time Systems	322
10.2.1 Obtaining Output of the System	327
10.3 Problems	328
A Laurent Series	331
A.1 Brief Review of Complex Numbers	331
A.1.1 Arithmetic Operations	332
A.1.2 De Moivre's Formula	333
A.1.3 Roots of a Complex Number	334
A.2 Complex Plane	335
A.3 Complex Analysis	336
A.3.1 Continuous Complex Function: The Limit of a Complex Function	337
A.3.2 Differentiable Complex Function	337
A.3.3 Analytic Complex Function: Cauchy-Riemann Equations	337
A.4 Complex Integration: Line Integral	338
A.4.1 Complex Line Integral: Special Results	339
A.5 Cauchy Integral Theorem	340
A.6 Cauchy Integral Formula	340
A.7 Derivatives of Analytic Functions	340
A.8 Geometric Series	341
A.9 Taylor Series	341
A.10 Maclaurin Series	342
A.11 Laurent Series	342
A.11.1 Important Observation	343
A.12 Isolated Singular Points	344
A.13 Residues	344