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VoIP Voice and Fax Signal Processing

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This book is dedicated to

- ***VoIP*** *and* ***Signal Processing Contributors***
- my ***Teachers***

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Sivannarayana Nagireddi

ABOUT THE AUTHOR

Sivannarayana Nagireddi, PhD, is currently working as the architect of voice over IP solutions at Ikanos Communications, Inc., and leads DSP and VoIP team. Dr. Sivannarayana and his team developed VoIP solutions including signal processing algorithms for voice and fax enabled residential gateway processors, which have been deployed by telecommunications providers.

Sivannarayana has been working on digital signal processing and systems for the last 22 years. His contributions in voice and VoIP started in 1999 with Encore Software, India. In early 2000, he built a DSP team for voice applications for Chiplogic India, and later on by mid-2000, he started managing VoIP solutions for Chiplogic USA. During the merger of Chiplogic with Analog Devices, Inc., he continued his VoIP solutions effort for Analog Devices, Inc. After working for 5 years at Analog Devices, Inc., he moved to Ikanos Communications, Inc., at the time of the acquisition of the network processor and ADSL ASIC product lines from Analog Devices, Inc.

Prior to contributions into voice and VoIP applications, for about 13 years from 1986 to 1999, he was working on signal processing algorithms and building systems for communication, radars, image processing, and medical applications.

Sivannarayana graduated with a degree in engineering from the Institute of Electronics and Telecommunications Engineering (IETE), New Delhi, India, in 1985. He received a Masters degree in electronics and communications engineering (ECE) from Osmania University, India. He was then awarded the PhD from the ECE Department, Osmania University, with a focus on wavelet signal processing applications.

His favorite topics are time-frequency analysis and communication signal processing, as well as building complete systems and supporting them for successful use. He is a member of the IEEE, a Fellow of IETE-India, and a reviewer for *Medical Engineering & Physics Journal* (Elsevier-UK).

PREFACE

Voice over IP (VoIP) gained popularity through actual deployments and by making use of VoIP-based telephone and fax calls with global roaming and connectivity via the Internet. Several decades of effort have gone into VoIP, and these efforts are benefitting real applications. Several valuable books have been published by experts in the field. While I was building the team, and training them, and conducting several design and support phases, I felt like a consolidated view and material on VoIP voice and fax signal processing was missing. Several contributions in the form of white papers, application notes, data sheets, standards, several books at the system level, and specialized books on signaling, speech compression, echo cancellation, and voice quality exist. Fax processing is available in books mainly for a public switched telephone network (PSTN), several white papers on fax over IP (FoIP), and a lot of ITU recommendations.

In this book, I am trying to bring out a consolidated view and basic approach with interpretation on popularly used techniques mapped to VoIP voice and fax signal processing. As a summary, this book broadly covers topics such as PSTN and VoIP overview, VoIP infrastructure, voice interfaces, voice signal processing modules and practical aspects, wideband voice, packetization, voice bit rate on multiple network interfaces, testing at module level and as a total VoIP system, fax on PSTN, FoIP processing, FoIP anomalies, testing, FoIP bit rates, miscellaneous topics that include country-specific deviations, bandwidth issues, voice quality improvements, processors and OS, and FAQs on VoIP and FoIP.

This book is organized into 22 chapters. In Chapter 1, PSTN interfaces, transmission requirements, as well as power and quantization levels are presented to create continuity for the subsequent chapters. In Chapter 2, connectivity between PSTN and VoIP, VoIP infrastructure and their architectures, pictures and interfaces of some of the practically deployed boxes, and their functions are presented. Software at block level for voice and fax, acoustic and network interfaces, VoIP signaling, and end-to-end VoIP call flow are also given in this chapter. Even though the first two chapters are introductory, several concepts required for subsequent chapters are systematically presented.

In Chapter 3, the popular voice compression codecs considered for VoIP deployment and their voice quality considerations one presented. Chapter 4 is on VAD/CNG for saving Internet bandwidth. Various inter-operation issues and testing is also given in this chapter. Chapter 5 is on packet loss concealment that improves voice quality in packet loss conditions. These three chapters are presented in a row to deal with voice compression and its extensions. Required overview on software, testing, complexity, quality, and their dependencies are also presented in these three chapters.

Echo cancellation is a big topic with several books exclusively written on that topic. I covered in Chapter 6 concepts mapped to telephones, telephone interfaces, VoIP CPE echo generation, rejection, and testing. DTMF is more of a time-frequency analysis problem with time sensitivity for generation, detection and rejection operations. In Chapter 7, a consolidated view of DTMF with illustrations and mathematical derivations for tones generation, detection, and rejection is given. Required emphasis on testing and country-specific deviations are also given in Chapter 7. As an extension on DTMF, Chapter 8 presents about different caller ID features that have close relations with basic tones, DTMF, phone and interfaces, various timing formats, caller ID and call progress tones detection, and working principles. Chapter 9 is on wideband voice with an example created using a VoIP adapter that addresses both narrow and wideband combinations. Wideband voice provides higher quality and is expected to be widely available in terminals such as IP phones, WiFi phones, and multimedia terminals.

Chapter 10 is on RTP, RTCP, packetization, packet impediments, and jitter buffers. On jitter buffers, several details are provided with illustrations, mathematical formulations, algorithms, various modes of operations, and helpful recommendations included. The VoIP bit rates from various codecs, network interfaces, and recommendations from practical deployments are given in Chapter 11. The network bit rate is usually given up to VoIP headers. In this book, interface headers, exact calculations, and tables with codec, packetization, and network interfaces are presented. Some clock options and interpretation of clock influences with simple calculations are given in Chapter 12. VoIP quality is influenced by the clock oscillator frequency and its stability. In Chapter 13, a high-level description of the VoIP voice tests and some of the instruments used for testing are presented.

Chapters 14–16 are dedicated to fax signal processing. In Chapter 14, a fax operation on PSTN, an end-to-end fax call, fax call phases, different fax call set-up tones, modulations, and demodulation schemes are presented that provide the background for FoIP. Chapter 15 is mainly on FoIP and gives an introduction to modem over IP at a high-level. The end-to-end VoIP fax call is given with SIP signaling in several diagrams for easy understanding of FoIP. The conditions for successful fax and modem calls and interoperability issues in FoIP are highlighted along with testing. A real-time VoIP fax is sent as a G.711 voice call or T.38 fax relay. In the literature, FoIP detailed bandwidth calculations are not listed. G.711 takes a lot of bit rate, whereas T.38 takes a small fraction of it. In Chapter 16, detailed headers and bandwidth calculations on Ethernet and DSL interfaces for various fax modulation rates and redundancy levels are given.

Similar to PSTN, VoIP has several dependencies for multiple country deployments that are discussed in Chapter 17. Each country and region has several deviations in its central office configurations, such as transmission lines, telephone impedances, tones, and acoustics. Chapter 18 is on IPQoS issues related to the bandlimited network, delay, and jitter for voice packets. Interpretation of the bandlimited nature, bandwidth, delay calculations, and recommendations for various packet sizes as a trade-off among packet sizes, delays, and fragmentation are given in this Chapter 18. The goal here is to improve the voice quality. Architectural, hardware processors, processing, and operating system considerations for VoIP are given in Chapter 19. Chapter 20 discusses consolidation of voice quality evaluation as well as various quality assessments through subjective, PESQ, and E-model. A list of major contributors of quality degradation and improvement options are included in this chapter.

Several questions and answers on voice and VoIP are provided in Chapter 21. About 100 questions and answers are given that systematically cover the topics listed in this book and are supplemented with several points that could not be directly addressed in continuity. Similarly, a fax FAQ section is given in Chapter 22. My expectation is that a sequential reading of these fax FAQs will give a quick overview of the fax processing flow in PSTN and FoIP.

The algorithms and mathematics are made fairly simple like arithmetic, and they are supplemented with several illustrations, direct results in tables, and summaries or recommendations on various aspects. Several FAQs in Chapters 21 and 22 will help for easy reading of the book. I tried to make this book simple to understand by many readers across several roles. I hope this book will help in understanding voice and fax signal processing for many new engineers, new contributors of VoIP, and students at the graduate and postgraduate level, as well as for managers, business, sales, and marketing teams, customers, and service providers.

In conclusion, several books are forthcoming that are going to address voice quality in general and wideband voice in particular. The contributions on wideband voice and signal processing techniques that are expected will create more natural conversation with a higher mean opinion score.

GLOSSARY

3GPP Third-generation partnership project

A Advantage factor (in R-factor)

AAL5 ATM adaptation layer 5

ABNF augmented Backus–Naur form

AC alternating current

ACELP algebraic code excited linear prediction

ACK acknowledgment

ACR absolute category rating

ADC analog-to-digital converter

ADPCM adaptive differential pulse code modulation

ADSL asymmetric DSL

ADSL2 asymmetric DSL 2

AFE analog front end

AGC automatic gain control

AJB adaptive jitter buffer

A-law logarithmic 64-kbps compression, which is the same as G.711 PCMU

ALC automatic level control

ALG application level gateway

ALU arithmetic logic unit (ALU)

AM amplitude modulation

AMR adaptive multi rate

AMR-HR AMR half rate

AMR-FR AMR full rate

AMR-NB adaptive multirate narrowband

AMR-WB adaptive multirate wideband

ANS answer tone, which is the same as CED

/ANS ANS with phase modulation

ANSam ANS tone with amplitude modulation

/ANSam ANS tone with amplitude and phase modulation
ANSI American National Standards Institute
APP application-specific function
ARQ automatic repeat request
ASN abstract syntax notation
ASN.1 Abstract syntax notation.1
ATM asynchronous transfer mode
ATT American Telephone and Telegraph

BCG bulk call generator
B-Channel Bearer Channel
BNLMS block normalized least mean square
BORSHT battery, overvoltage protection, ringing, supervision, hybrid, and test functions (in the telephone interface)
BPF band-pass filter
BPI baseline privacy interface
BPSK binary phase-shift keying
BRI basic rate interface
BT British Telecom
BurstR burst ratio
BW bandwidth
Byte or byte 8-bits of data

CA call agent
CAR receiving terminal activation signal (Japan-caller ID)
CAS CPE alerting signal
CAS channel-associated signaling
CC CSRC count
CCA Cable Communications Association
CCITT Committee Consultative International Telegraph and Telephone
CCR comparison category rating
CED called terminal identification tone
CELP code excited linear prediction
CFR confirmation to receive
CID caller identity delivery or caller ID
CIDCW calling identity delivery on call waiting or caller ID on call waiting
CI call indication
CJ CM terminator
CLASS custom local area signaling services

CLI caller line identification

CLIP caller line identity presentation

CLIR caller line identification restriction

CLR circuit loudness rating

CM call menu

CM cable modem

CMOS comparison mean opinion score

CMTS cable modem terminal system

CND calling number display (on CPE)

CND calling number delivery (on CO)

CN comfort noise

CNG calling tone in fax call

CNG comfort noise generation

CO central office

codec voice coder (compression) and decoder (decompression) (in this book)

CODEC COder (hardware ADC) and DECoder (hardware DAC) or SLAC (in this book)

Coef coefficient

Compander compressor and expander

Cos(...) cosine function

CP call progress

CPE customer premises equipment

CPI common part indicator

CPTD call progress tone detection

CPTG call progress tone generation

CPU central processing unit

CRC cyclic redundancy check

CRLF carriage return line feed

CRP command repeat

CS-ACELP conjugate-structure algebraic-code-excited linear-prediction

CSI called subscriber identification

CRLF carriage return line feed

CSeq command sequence

CSRC contributing sources

CT call tone

CTC continue to correct

CTR continue to correct response

DA destination address

DAA digital access arrangement

DAC digital-to-analog converter

dB deciBel

dBm decibel power with 1 milliWatt reference power

dBm0 dBm of the signal that would be measured at the relevant 0-dBr level reference point

dBov dB relative to the overload point of the digital system

dBr power with zero-level point (used to refer to relative power level)

dBrnc noise power with 1 picoWatt reference and c-message filter weighting

dBp noise power with psophometric weighting

dBSPL The sound pressure with 20 μPa (microPascal) as reference

dBV RMS voltage in dB with 1-V RMS as reference

D-Channel Data channel

DC direct current

DCE data communications equipment

DCME digital circuit multiplication equipment

DCT discrete cosine transform

DCN disconnect

DCR degradation category rating

DCS digital command signal

DDR double data rate (memory)

DECT digital enhanced cordless telecommunications

DESA discrete energy separation algorithm

DFT discrete Fourier transforms

DIS digital identification signal

DLC digital loop carrier

DM data memory (in processors)

DMA direct memory access

DMIPS Dhrystone MIPS

DMOS degradation mean opinion score

DOCSIS data over cable service interface specifications

dpi dots per inch

DS digital signaling

DS3 digital Service, Level 3

DSL digital subscriber line

DSLA digital speech level analyzer

DSLAM DSL access multiplexer (central office equipment for DSL service)

DSP digital signal processor

DT double talk

DTC digital transmit command

DTD double-talk detector

DT-AS dual-tone alerting signal

DTE data terminal equipment

DTMF dual-tone multifrequency

DTX discontinuous transmission

E1 E-carrier digital signaling

E-model Electrical-model

EBI even bits inversion

EBIU extended bus interface unit

EC echo canceller

ECM error correction mode

EN enterprise networks

EOL end of line

EOM end of message

EOP end of procedure

EOR end of retransmission

ERL echo return loss

ERLE echo return loss enhancement

ERR end of retransmission response

ETSI European Telecommunications Standards Institute

EV embedded variable

Fax facsimile (Facsimile meaning "a copy")

FaxLab fax testing instrument from Qualitylogic

FCD facsimile-coded data

FCF facsimile control field

FCS frame check sequence

FDM file diagnostic message

FEC forward error correction

FFT fast Fourier transform

FGPS physical layer overhead F—FEC, G—Guard Time, P—Preamble, S—Stuffing bytes

FIF facsimile information field

FIR finite impulse response
FJB fixed jitter buffer
FM frequency modulation
FMC fixed mobile convergence
FoIP fax over IP
FOM figure of merit
FSK frequency-shift keying
FT French Telecom
FTT fail to train
FXO foreign exchange office
FXS foreign exchange subscriber or station

G1 Group-1 facsimile
G3 Group-2 facsimile
G3 Group-3 facsimile
G3C Group 3C facsimile
G3FE Group-3 facsimile equipment
G4 Group-4 facsimile
G711WB wideband embedded extension for G.711 PCM
GDMF Generic data message format
GIPS Global IP sound
GoB Good or better
GPS Global positioning system
GR General requirements
GSM Global system for mobile communications
GUI Graphic user interface
GW Gateway

H registers echo canceller filter memory
HCS header check sum
HDLC high-level data link control
HEC header error control
HG home gateway (CPE)
HPF High-pass filter
HTTP Hypertext transfer protocol
Hz Hertz, frequency in cycles per second

IAD integrated access device
IAF Internet-aware fax device