
Beyond VoIP Protocols

**Understanding Voice Technology
and Networking Techniques for IP Telephony**

IP Telephony: two-book reference set

Beyond VoIP Protocols: Understanding Voice Technology and Networking Techniques for IP Telephony is a companion reference to *IP Telephony: Deploying Voice-over-IP Protocols*. More details of this companion text may be found on the last page of this book.

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**Understanding Voice Technology
and Networking Techniques for IP Telephony**

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Jossey-Bass, 989 Market Street, San Francisco, CA 94103-1741, USA

Wiley-VCH Verlag GmbH, Boschstr. 12, D-69469 Weinheim, Germany

John Wiley & Sons Australia Ltd, 33 Park Road, Milton, Queensland 4064, Australia

John Wiley & Sons (Asia) Pte Ltd, 2 Clementi Loop #02-01, Jin Xing Distripark, Singapore 129809

John Wiley & Sons Canada Ltd, 22 Worcester Road, Etobicoke, Ontario, Canada M9W 1L1

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British Library Cataloguing in Publication Data

A catalogue record for this book is available from the British Library

ISBN 0-470-02362-7

Typeset in 10/12pt Times by Laserwords Private Limited, Chennai, India

Printed and bound in Great Britain by TJ International, Padstow, Cornwall

This book is printed on acid-free paper responsibly manufactured from sustainable forestry in which at least two trees are planted for each one used for paper production.

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Abbreviations

%GoB	Percent good or bad
%PoW	Percent poor or worse
3GPP	3rd Generation Partnership Project
A/D	Analog to digital
AAC	Advanced audio coding
AAL5	ATM Adaptation Layer 5
AAP	Multicast Address Allocation Protocol (draft-handley-aap-00.txt)
ABR	Available bitrate; area border router
ABS	Analysis by synthesis
ACELP	Algebraic code excited linear prediction
ACR	Absolute category rating; subjective test (MOS)
ADM	Adaptive delta modulation (modulation delta)
ADPCM	Adaptive differential pulse code modulation (MICDA)
ADSL	Asymmetric Digital Subscriber Line
ADSPEC	ADvertisement Specification (RSVP)
AEC	Acoustic echo cancelation
AMPS	Analog mobile phone standard
AMR	Adaptive multi-rate
AMR-WB	Adaptive multi-rate (wide band)
ANSI	American National Standard Institute
API	Application Programming Interface
ARIB	Association of Radio Industries and Businesses
ASBR	AS boundary router
ASVD	Analog simultaneous voice and data
ATC	Adaptive transform coding
ATM	Asynchronous Transfer Mode
BA	Behavior aggregate

BDR	Backup designated router
BER	Bit error rate
BFI	Bad frame indicator
BGMP	Border Gateway Multicast-routing Protocol.
C/D	Continuous to discrete
CAPEX	Capital Expenditure
CAT	Client accept
CBQ	Class-based queuing
CBR	Constant bitrate
CBT	Core-based tree
CC	Client close
CCR	Comparison category rating (subjective tests)
CDMA	Code division multiplex access
CDSL	Consumer Digital Subscriber Line
CELP	Code-excited linear prediction (vector quantization of excitation)
CGMP	Cisco Group Management Protocol
CID	Context identifier
CIDR	Classless Interdomain Routing
CLP	Cell loss priority
CLR	Circuit loudness rating
CM	Cable modem
CMOS	Comparison mean opinion score
CMS	Call management server
CMTS	Cable modem termination system
CNAME	Canonical Name
CNG	Comfort noise generation
COPS	Common Open Policy Service
CoS	Class of service
CPE	Customer Premises Equipment
CPU	Central processing unit
CRC	Cyclic redundancy check
CRCX	Create Connection
CRTTP	Compressed Real Transport Protocol
CS-ACELP	Conjugate-structure ACELP
CT	Cordless telecommunications
CT2	CT system two
CU	Currently unused
CWTS	China Wireless Telecommunication Standard group
D/A	Digital to analog
D/C	Discrete to continuous
DAMPS	Digital AMPS (Analog Mobile Phone Service)
DCME	Digital circuit multiplication equipment
DCR	Degradation category rating test (subjective test: DMOS)

DCS	Distributed call signaling
DCT	Discrete cosine transform
DE	Discard eligible bit
DEC	Decision
DECT	Digital Enhanced cordless telephone
DMOS	Degradation mean opinion score
DNS	Domain name system
DoD	Department of Defense (US)
DQoS	Dynamic quality of service
DR	Designated router
DRQ	Delete request state
DS	Differentiated service(s)
DSCP	Differentiated services codepoint
DSL	Digital Subscriber Line
DSLAM	DSL Multiplexer
DSP	Digital signal processor (fixed point or floating point)
DSVD	Digital simultaneous voice and data
DTMF	Digital tone multi-frequency
DTX	Discontinuous transmission
DVMRP	Distance Vector Multicast-routing Protocol
EDGE	Enhanced Data Rates for GSM Evolution
EEC	Electric echo canceler
EFMA	Ethernet First Mile Association
EFR	Enhanced full rate
EGP	Exterior Gateway Protocol
EMTA	Embedded multimedia terminal adapter
ERLE	Echo return loss enhancement
ETSI	European Telecommunications Standardization Institute
FCC	Federal Communication Commission
FCFS	First come first served
FEC	Forward error correction
FER	Frame error rate
FIB	Forwarding information base, or forwarding table
FIFO	First in first out
FIR	Finite impulse response
FPLMTS	Future public land mobile telecommunication system
FR	Full rate
FS1015	Federal Standard 1015: 2.4-kbit/s LPC speech coder (NATO)
FS1016	Federal Standard 1016: 4.8-kbit/s CELP speech coder (NATO)
GARP	Generic Attribute Registration Protocol (formerly Group Address Resolution Protocol)
GC	Gate controller
GMRP	GARP Multicast Registration Protocol
GPS	Generalized processor sharing
GSM	Global System for Mobile communications

GSM-EFR	GSM-enhanced full-rate speech coder 13 kbit/s-NPAG
GSM-FR	GSM full-rate RPE-LTP 13 kbit/s
GSM-HR	GSM half-rate VSELP 5.6 kbit/s
GSTN	General switch telephone network (deregulated PSTN!)
HDLc	High-Level Data Link Control
HDVMP	Hierarchical DVMP
HTTP	Hypertext Transfer Protocol
IANA	Internet Assigned Numbers Authority
ICMP	Internet Control Message Protocol
IGMP	Internet Group Membership Protocol
iif	Incoming interface
IIR	Infinite impulse response
IP	Internet Protocol
IRC	Internet Relay Chat (protocol)
IS-xx	Intermediate standard xx (cf. IS-54 VSELP)
ISDN	Integrated service digital network (RNIS, NUMERIS)
ISO	International Standardization Organization
ISP	Internet service provider
ITU	International Telecommunications Union
JB	Jitter buffer
JND	Just noticeable distortion
KA	Keep alive
L2TP	Layer 2 Tunneling Protocol
LAN	Local area network
LARs	Logarithmic area ratios
LD-CELP	Low-delay CELP (ITU-T G.7728)
LFI	Link fragmentation and interleaving
LMS	Least mean squares
log-PCM	Logarithmic pulse code modulation (G.711 A-law or μ -law)
LPC	Linear predictive coding; linear prediction coefficient
LR	Loudness rating
LSA	Link-state advertisement
LSP	Line spectral pair
LTP	Long-term prediction
MA	Moving average
MAAS	Multicast address allocation server
MAC	Medium Access Control (Layer)
MARS	Multicast Address Resolution Server
MASC	Multicast Address Set Claim protocol (draft-ietf-idmr-masc-00.txt)
MBE	Multi-band excitation
MBR	Multicast border router
MBZ	Must be zero
MCML	Multi-class extension to multilink PPP
MCU	Multipoint control unit

MDCX	Modify Connection
MDHCP	Multicast Dynamic Host Configuration Protocol (address allocation extensions)
MEF	Metro Ethernet Forum
MELP	Mixed excitation linear predictor
MF	Multi-field
MGCP	Media Gateway Control Protocol
MIPS	Millions of instructions per second
ML-PPP	Multilink PPP
MLT	Modulated lapped transform
MNRU	Modulated noise reference unit
MOS	Mean opinion score (subjective tests)
MOS _{CQE}	Mean opinion score, conversational quality evaluation
MOSPF	Multicast extension to OSPF
MP-MLQ	Multipulse maximum likelihood quantization (G.723.1 at 6.3 kbit/s)
MPEG	Moving Picture Expert Group
MPLS	Multiprotocol label switching
MSB	Most Significant Bit
MSC	Mobile switching center
MSE	Mean Square Error
MTA	Multimedia terminal adapter
MTU	Maximum transmission unit
NBMA	Non-broadcast multiple access
NCS	Network-based call signaling
NICAM	Near-instantaneous companding and multiplexing
NLRI	Network-layer reachability information
NMR	Noise-to-mask ratio
NNTP	Network News Transfer Protocol
NPAG	North American PCS 1900 Action Group
NTP	Network Time Protocol
OLR	Overall loudness rating
OPEX	Operational Expenditure
OPN	Client open
OSPF	Open Short Path First (Protocol)
PAL	Phase Alternation by Line (TV standard)
PAM	Pulse amplitude modulation
PARCORS	PARTial CORrelatorS
PASTA	Poisson Arrivals See Time Averages
PBX	Private branch exchange
PCM	Pulse code modulation (MIC)
PCME	Packet circuit multiplication equipment
PCN	Personal communication network
PCS	Personal communication system
PDC	Personal digital communication

PDCP	Packet Data Convergence Protocol
PDF	Probability density function
PDP	Policy decision point
PEP	Policy enforcement point
PGPS	Packet-generalized processor sharing
PHB	Per-hop behavior
PHS	Personal handy-phone system
PID	Protocol ID
PIM	Protocol-independent multicast
PIM-SM	PIM (sparse mode)
PIM-DM	PIM (dense mode)
POP	Point of Presence
POTS	Plain Old Telephone Service
PPP	Point to Point control Protocol
PPP-ML	Multilink PPP
PSI-CELP	Pitch Synchronous Innovation CELP
PSTN	Public-switched telephone network
PT	Payload Type
QDU	Quantization distortion unit
QMF	Quadratic mirror filter
QoS	Quality of service
RADSL	Rate Adaptative DSL
RAM	Random access memory
RED	Random early detection
RELp	Residual excited linear prediction
REQ	Request
RFC	Request for comments
RIB	Routing information base
RIP	Routing Information Protocol
RLC	Running length code
RLR	Receive loudness rating
ROHC	Robust header compression
ROM	Read-only memory
RP	Rendezvous point
RPB	Reverse path broadcasting
RPE-LTP	Regular pulse-excited LPC with long-term prediction
RPF	Reverse path forwarding
RPM	Reverse path multicasting
RPT	Report state
RQNT	Request Notification
RSP	Route switch processor (Cisco)
RSRR	Routing support for resource reservation interface
RSVP	Resource ReserVation Protocol
RTCP	Real-time Transport Control Protocol
RTP	Real time Transport Protocol

RTP-Mux	Multiplex RTP
SAP	Session Announcement Protocol
SB-ADPCM	Subband ADPCM (ITU-T G.722)
SBC	Subband coding
SCFQ	Self-clocked fair queuing
SDH	Synchronous Digital Hierarchy
SDP	Session Description Protocol
SDR	Session directory
SDSL	Symmetric DSL
SECAM	Système Electronique Couleur avec Mémoire (TV Standard)
SHDSL	Single-Pair High-Speed Digital Subscriber Line
SID	Silence description
SIP	Session Initiation Protocol
SLA	Service-level agreement
SLR	Send loudness rating
SMR	Signal-to-mask ratio
SNA	System Network Architecture
SNR	Signal-to-noise ratio
SPL	Sound pressure level
SSC	Synchronize complete
SSQ	Synchronize state request
SSRC	Synchronisation Source (Identifier)
STC	Sinusoidal transform coder
TCA	Traffic-conditioning agreement
TCL	Terminal coupling loss
TCLwdt	Weighted terminal coupling loss—double talk
TCLwst	Weighted terminal coupling loss—single talk
TCP	Transport Control Protocol
TCRTP	Tunneling-multiplexed Compressed RTP
TDM	Time division multiplexing
TDMA	Time division multiplex access
TELRL	Talker echo loudness rating
TIA	Telecommunications Industry Association (USA)
TOS	Type of service
TRAU	Transcoder/Rate Adaptation Unit
TRPB	Truncated reverse path broadcasting
TSPEC	Traffic Specification
TTA	Telecommunications Technologies Association
TTC	Telecommunication Technologies Committee
TTL	Time to live
UADSL	Universal Asymmetric Digital Subscriber Line
UBR	Unspecified Bitrate
UDP	User Datagram Protocol
UDSL	Universal Digital Subscriber Line

UED/UEP	Unequal bit error detection and protection
UMTS	Universal Mobile Telecommunication System
URL	Uniform (universal) resource locator
USB	Universal Serial Bus
UTRAN	Universal Terrestrial Radio Access Network
VAD	Voice activity detector
VAT	Visual Audio Tools
VC	Virtual channel
VDSL	Very High Bitrate Digital Subscriber Line
VIC	mBone “Video Conferencing” tool
VIP	Versatile interface processor (Cisco)
VLAN	Virtual LAN
VLSI	Very large scale integration
VoIP	Voice over Internet Protocol
VQ	Vector quantization
VSELP	Vector sum-excited linear prediction
WAN	Wide-area network
WB-AMR	Wideband AMR
WFQ	Weighted fair queuing
WiFi	Wireless Fidelity
WRED	Weighted random early detection
WSNR	Weighted SNR (signal-to-noise ratio)
xDSL	Any type of DSL technology

Glossary

Aggregate *See* ‘Behavior aggregate’.

BA classifier A traffic classifier based on the DS field.

Behavior aggregate DiffServ term defined in RFC 2474 as ‘a collection of packets with the same codepoint crossing a link in a particular direction’.

Boundary link A link connecting the edge nodes of two domains (RFC 2475).

Boundary node A DS node that connects one DS domain to a node either in another DS domain or in a domain that is not DS-capable (RFC 2475).

Circuit loudness rating (CLR) Loudness loss between two electrical interfaces in a connection or circuit, each interface terminated by its nominal impedance which can be a complex value. This is 0 for a digital circuit and 0.5 for a mixed analog/digital circuit.

Class selector codepoint DiffServ term defined in RFC 2474 as ‘any of the eight codepoints in the range xxx000’ (x = 0 or 1). *See also* ‘Class selector-compliant codepoint’.

Class selector-compliant codepoint DiffServ term defined in RFC 2474 as per-hop behavior satisfying the class selector specifications as defined in RFC 2474. In short, these requirements aim at ensuring a minimal level of backward compatibility with IP precedence semantics of RFC 791 (see Chapter 4 on QoS for more details).

Codepoint Proposed name for the value of the PHB field of the DS octet, in the DiffServ framework (*see* RFC 2474 and class selector codepoint).

Controlled load service An application requesting a controlled load service for a stream of given characteristics expects the network to behave as if it was lightly loaded for that stream.

Currently unused (CU) The last two bits of the DS octet.

dBm Power level with reference to 1 mW.

dBm0 At the reference frequency (1,020 Hz), L dBm0 represents an absolute power level of L dBm measured at the transmission reference point (0-dBr point), and a level of $L + x$ dBm measured at a point having a relative level of x dBr (see G.100, annex A.4).

Differentiated service(s) (DS) The new name assigned by the IETF DiffServ group to the Ipv4 TOS field and the IPv6 traffic class field (*see* RFC 2474).

Differentiated services codepoint (DSCP) The name of the first 6 bits of the DS octet (in drafts before RFC 2474 these bits were called the PHB).

DS-compliant Behaving according to the general rules of RFC 2474 (*see* DS).

Echo Unwanted signal delayed to such a degree that it is perceived as distinct from the wanted signal.

Exterior gateway protocol (EGP) Used for unicast interdomain routing (e.g., BGP).

First come first served (FCFS) Another name for FIFO.

First in first out (FIFO) Same as FCFS.

Forwarding table For unicast routers, this is the list of the appropriate egress interface for each destination prefix. For a multicast router, this also includes the expected incoming interface (iif) and a list of outgoing interfaces (oiflist) for each destination group address (there can only be one such entry for each source for some multicast-routing protocols, like DVMRP).

Hierarchical DVMRP (HDVMP) *See* A.S. Thyagarajan and S.E. Deering. In: *Proceedings of the ACM SIGCOMM* Hierarchical distance-vector multicast routing for the MBone, pp. 60–66, October 1995.

In profile Packets part of a packet stream that were found to comply with the packet stream description (average and peak rate, maximum burst size, etc. ...).

Listener echo Echo produced by double-reflected signals and disturbing the listener.

Loudness rating (LR) As used in the G-Series Recommendations for planning, loudness rating is an objective measure of loudness loss (i.e., weighted, electro-acoustic loss between certain interfaces in the telephone network). If the circuit between the interfaces is subdivided into sections, the sum of individual section LRs is equal to the total LR. In LR contexts, the subscribers are represented from a measuring point of view by an artificial mouth and an artificial ear, respectively, both being accurately specified.

Meter A device that performs metering (RFC 2475).

Metering The process of measuring the temporal properties (e.g., rate) of a traffic stream selected by a classifier. The instantaneous state of this process may be used to affect the operation of a marker, shaper, and dropper, and/or may be used for accounting and measurement purposes (RFC 2475).

MF classifier A traffic classifier based on one or more IP header fields such as protocol number, source and destination IP addresses, port numbers, DS field value, etc. (*see also* BA classifier and the next entry).

MF classifier: multi-field classifier (MF) A functional block that is able to sort flows according to several fields of the IP packet (source address, destination address, source port, destination port, ...).

Microflow A single instance of an application-to-application flow of packets which is identified by source address, source port, destination address, destination port, and/or protocol ID (*see also* MF classifier) (RFC 2475).

Multicast RIB The routing information base, or routing table, used to calculate the 'next hop' toward a particular address for multicast traffic.

Network layer reachability information (NLRI) Conveyed by BGP4+, this information is used by BGMP to inject multicast routes in the interdomain-routing protocol.

oiflist A list of outgoing interfaces which is part of each forwarding table entry.

Out of profile Property of data packets within a flow which momentarily exceeds some envelope parameters of its profile (such as maximum burst size) (e.g., if the flow is regulated by a token bucket, packets arriving when there are no tokens and the backlog buffer is full are out of profile) (*see* Profile).

Overall loudness rating (OLR) Loudness loss between the speaking subscriber's mouth and the listening subscriber's ear via a connection.

PHB group A set of one or more PHBs that can only be meaningfully specified and implemented simultaneously, due to a common constraint applying to all PHBs in the set such as queue servicing or queue management policy (RFC 2475).

PHB Defined in RFC 2474 as 'a description of the externally observable forwarding treatment applied at a differentiated services compliant node to a behavior aggregate'. PHB also referred to the first six bits of the DS octet in drafts before RFC 2474 (these bits are now called DSCP).

Policing The process of discarding packets by a dropper within a traffic stream in accordance with the state of a corresponding meter enforcing a traffic profile (RFC 2475).

Profile Properties of a data flow, usually defined as envelope parameters (such as maximum burst size) and mean values (such as average bitrate).

Promiscuous An interface set in promiscuous mode receives and forwards to upper layers (the device driver), all the packets it has access to, even if the physical destination address of such packets shows it is destined to another interface.

Prune A message sent by a downstream multicast router to an upstream router, meaning he is not interested in receiving multicast packets for a specific group and source. This marks a soft state in the upstream router, which usually expires after an hour or two.

Receive loudness rating (RLR) Loudness loss between an electric interface in the network and the listening subscriber's ear. Loudness loss is here defined as the weighted (dB) average of driving e.m.f. to measured sound pressure. The weighted mean value for G.111 and G.121 is 1–6 in the short term and 1–3 in the long term (from G.111).

Routing information base (RIB) The list of all routes (next hop and distance to each destination prefix) from the router.

Send loudness rating (SLR) Loudness loss between the speaking subscriber's mouth and an electric interface in the network. Loudness loss is here defined as the weighted (dB) average of driving sound pressure to measured voltage. The weighted mean value for G.111 and G.121 is 7–15 in the short term and 7–9 in the long term (from G.111).

Soft state Any state that times out after a certain delay if not refreshed.

Source router In this document only: any router directly connected to a subnetwork with a source station.

Stub domain A domain that has no transit traffic between its border routers (i.e., not used by other domains as a transit domain to destinations external to the domain).

Talker echo loudness rating (TELRL) Loudness loss of the sound of the speaker's voice reaching his ear as a delayed echo (*see* 4.2/G.122 and fig. 2/G.131).

Talker echo Echo produced by reflection near the listener's end of a connection and disturbing the talker.

Terminal coupling loss (TCL) Coupling loss between the receiving port and the sending port of a terminal due to acoustical coupling at the user interface, electrical coupling due to crosstalk in the handset cord or within the electrical circuits, and seismic coupling through the mechanical parts of the terminal. For a digital handset it is commonly in the order of 40 dB to 46 dB.

Traffic-conditioning agreement (TCA) The specification of all traffic-shaping parameters, discard policies, in/out-of-profile handling rules used for a particular service-level agreement (SLA).

Transit domain A domain that has transit traffic between its border routers (i.e., used by other domains to reach destinations external to the domain).

Weighted terminal coupling loss—double talk (TCLwdt) The weighted loss between R_{in} and S_{out} network interfaces when echo control is in normal operation and when the local user and the far-end user talk simultaneously.

Weighted terminal coupling loss—single talk (TCLwst) The weighted loss between R_{in} and S_{out} network interfaces when echo control is in normal operation and when there is no signal coming from the user.