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Contents

Foreword

This Technical Specification has been produced by the 3rd Generation Partnership Project (3GPP).

The present document introduces the set of default codecs applying to 3G packet switched conversational multimedia applications within the 3GPP system.

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of the present document, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version x.y.z

where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

3 or greater indicates TSG approved document under change control.

y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

z the third digit is incremented when editorial only changes have been incorporated in the document.

Introduction

The present document contains a specification for default multimedia codecs to be used within 3GPP specified IP Multimedia Subsystem (IM Subsystem). IM Subsystem as a subsystem includes specifically the conversational IP multimedia services, whose service architecture, call control and media capability control procedures have been defined in 3GPP specifications TS 24.229 [15], and are based on the 3GPP adopted version of IETF Session Initiated Protocol (SIP).

The term codec is usually associated with a single media type. In case of packet switched transport domain, which IM Subsystem will depend on, the individual media types are independently encoded and packetised to appropriate separate Real Time Protocol (RTP) packets. These packets are then transported end-to-end inside UDP datagrams over real-time IP connections that have been negotiated and opened between the terminals during the SIP call as specified in 3GPP TS 24.229 [15].

From the codec definition viewpoint, the UEs operating within IM Subsystem need to provide encoding/decoding of the derived codecs, and perform corresponding packetisation/depacketisation functions. Logical bound between the media streams is handled in the SIP session layer, and inter-media synchronisation in the receiver is handled with the use of RTP time stamps.

Finally, since 3GPP networks are inherently error prone, error detection and/or correction must also be provided by the individual codecs within IM Subsystem, since they have a comprehensive view of the bit stream they produce and therefore can apply the most efficient form of error detection and/or correction.

1 Scope

The present document introduces the set of default codecs for packet switched conversational multimedia applications within 3GPP IP Multimedia Subsystem. Visual and sound communication are specifically addressed. The intended applications are assumed to require low-delay, real-time functionality.

The present document is applicable, but not limited, to services such as PS video telephony and Push to talk over Cellular (PoC) as well as Combined CS and IMS services (CSI). For Multimedia Telepphony Service for IMS (MTSI), all media handling, including codecs, is defined in TS 26.114 [50]. This specification does not apply for MTSI.

The applicability of this specification to GERAN is FFS.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

[1] IETF RFC 3261: "SIP: Session Initiation Protocol".

[2] IETF RFC 4566: "SDP: Session Description Protocol".

[3] IETF RFC 2429: "RTP Payload Format for the 1998 Version of ITU-T Rec. H.263 Video (H.263+)".

[4] IETF RFC 3550: "RTP: A Transport Protocol for Real-Time Applications", Schulzrinne H. et al, July 2003.

[5] IETF RFC 3016: "RTP Payload Format for MPEG-4 Audio/Visual Streams".

[6] ITU-T Recommendation H.263 (02/98): "Video coding for low bit rate communication".

[7] 3GPP TS 26.110: "Codec for Circuit Switched Multimedia Telephony Service; General Description".

[8] 3GPP TS 26.111: "Codec for Circuit Switched Multimedia Telephony Service; Modifications to H.324".

[9] 3GPP TS 26.071: "Mandatory Speech Codec speech processing functions; AMR Speech Codec; General description".

[10] 3GPP TS 26.090: "Mandatory Speech Codec speech processing functions; AMR Speech Codec; Transcoding functions".

[11] 3GPP TS 26.073: "Adaptive Multi-Rate (AMR); ANSI C source code".

[12] 3GPP TS 26.104: "ANSI-C code for the floating-point AMR speech codec".

[13] ISO/IEC 14496-2 (2004): "Information technology - Coding of audio-visual objects - Part 2: Visual".

[14] (void)

[15] 3GPP TS 24.229: "IP Multimedia Call Control Protocol based on SIP and SDP".

[16] 3GPP TS 26.171 (Release 5): "AMR speech codec, wideband; General description".

[17] 3GPP TS 26.190 (Release 5): "Mandatory Speech Codec speech processing functions AMR Wideband speech codec; Transcoding functions".

[18] 3GPP TS 26.201 (Release 5): "AMR speech codec, wideband; Frame structure".

[19] ITU-T Recommendation H.263 – Annex X (03/04): "Annex X: Profiles and levels definition".

[20] 3GPP TS 23.228: "IP multimedia subsystem; stage 2".

[21] 3GPP TS 23.107: "QoS Concept and Architecture".

[22] 3GPP TS 23.207: "End to end quality of service concept and architecture".

[23] 3GPP TS 23.060: "General Packet Radio Service (GPRS); Service description; Stage 2".

[24] Void

[25] ITU-T Recommendation T.140 (1998): "Protocol for multimedia application text conversation" (with amendment 2000).

[26] 3GPP TS 26.101: "Mandatory Speech Codec speech processing functions; AMR Speech Codec; Frame Structure".

[27] IETF RFC 2119: "Key words for use in RFCs to Indicate Requirement Levels".

[28] 3GPP TS 26.093: "Mandatory Speech Codec speech processing functions; AMR Speech Codec; Source Controlled Rate operation".

[29] 3GPP TS 46.060: "Enhanced Full Rate (EFR) speech transcoding".

[30] TIA/EIA -136-Rev.A, part 410 - "TDMA Cellular/PCS – Radio Interface, Enhanced Full Rate Voice Codec (ACELP). Formerly IS-641. TIA published standard, 1998".

[31] ARIB, RCR STD-27H, "Personal Digital Cellular Telecommunication System RCR Standard".

[32] IETF draft-westberg-realtime-cellular-01.txt, "Realtime Traffic over Cellular Access Networks".

[33] IETF draft-larzon-udplite-03.txt, "The UDP Lite Protocol".

[34] 3GPP TS 26.092: "Mandatory Speech Codec speech processing functions; AMR Speech Codec; Comfort noise aspects".

[35] IETF RFC 4867: "RTP payload format and file storage format for the Adaptive Multi-Rate (AMR) Adaptive Multi-Rate Wideband (AMR-WB) audio codecs", April 2007.

[36] IETF RFC 2833: “RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals”, May 2000.

[37] 3GPP TS 26.243: “ANSI C code for the Fixed-Point Distributed Speech Recognition Extended Advanced Front-end”.

[38] IETF RFC 4060 "RTP Payload Formats for European Telecommunications Standards Institute (ETSI) European Standard ES 202 050, ES 202 211, and ES 202 212 Distributed Speech Recognition Encoding".

[39] 3GPP TS 26.173: "ANCI-C code for the Adaptive Multi Rate - Wideband (AMR-WB) speech codec".

[40] 3GPP TS 26.204: "ANSI-C code for the Floating-point Adaptive Multi-Rate Wideband (AMR-WB) speech codec".

[41] ITU-T Recommendation H.264 (2003): "Advanced video coding for generic audiovisual services" | ISO/IEC 14496-10:2003: "Information technology – Coding of audio-visual objects – Part 10: Advanced Video Coding".

[42] ISO/IEC 14496-10/FDAM1: "AVC Fidelity Range Extensions".

[43] IETF RFC 3984 (2005): "RTP Payload Format for H.264 Video", S. Wenger, M.M. Hannuksela, T. Stockhammer, M. Westerlund and D. Singer.

[44] 3GPP TS 23.221: “Architectural requirements”.

[45] 3GPP TR 26.943: ”Recognition performance evaluations of codecs for Speech Enabled Services(SES); (Release 6)”.

[46] 3GPP TS 22.279: “Combining CS and IMS services; Stage 1”.

[47] 3GPP TS 23.279: “Combining CS and IMS services; Stage 2”.

[48] 3GPP TS 26.141: “IP Multimedia System (IMS) Messaging and Presence; Media formats and codecs”.

[49] IETF RFC 4103 "RTP payload for text conversation".

[50] 3GPP TS 26.114, “IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction”.

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply.:

3G PS multimedia terminal: terminal based on IETF SIP/SDP internet standards modified by 3GPP for purposes of 3GPP packet switched network based multimedia telephony

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

AMR	Adaptive MultiRate codec
AVC	Advanced Video Codec
CSI	Combination of CS and IMS services
DSR	Distributed Speech Recognition
IETF	Internet Engineering Task Force
IM Subsystem	Internet protocol Multimedia Subsystem
ITU-T	International Telecommunications Union-Telecommunications
MTSI	Multimedia Telephony Service for IMS
PoC	Push to talk over Cellular
RFC	IETF Request For Comments
RTCP	RTP Control Protocol
RTP	Real-time Transport Protocol
SDP	Session Description Protocol
SES	Speech Enabled Services
SIP	Session Initiated Protocol

4 General

3G PS multimedia terminals provide real-time video, audio, or data, in any combination, including none, over 3GPP IM Subsystem. Terminals are based on IETF defined multimedia protocols SIP, SDP, RTP and RTCP. Communication may be either 1-way or 2-way. Such terminals may be part of a portable device or integrated into an automobile or other non-fixed location device. They may also be fixed, stand-alone devices; for example, a video telephone or kiosk. Multimedia terminals may also be integrated into PCs and workstations.

In addition, interoperation with other types of multimedia telephone terminals, such as 3G-324M may be possible, however in such case a media gateway functionality supporting 3G-324M - IM Subsystem interworking will be required within or outside the IM subsystem.

For IMS terminals capable of Combined CS and IMS (CSI) operation [46][47], Annex A of specification [48] provides information on how to combine IMS media with CS calls.

5 System overview

The present document describes the required codec related elements for 3G PS multimedia terminal:

- codecs for 3G PS multimedia terminal;
- media encapsulation and decapsulation rules for each codec.

6 Functional requirements

SIP protocol itself does not mandate any codecs. Standardisation of codecs does not prevent the use of other codecs that can be signalled using the SDP protocol. 3G PS multimedia terminals shall be able to use the same audio and video codecs applied in 3G-324M [8]. This will ensure the interoperability with 3G circuit switched multimedia telephony.

6.1 Audio

3G PS multimedia terminals offering audio communication (including PoC services) shall support AMR narrowband speech codec [9], [10], [11] to [12].

The AMR wideband speech codec shall be supported when the 3G PS multimedia terminal supports wideband speech working at 16 kHz sampling frequency [16], [17], [39], [40].

The usage of telephone-event media format is recommended for DTMF.

Annex D provides guidelines for using audio in the context of PoC services.

6.2 Video

3G PS multimedia terminals offering video communication shall support ITU-T recommendation H.263 [6] [19] baseline (Profile 0) Level 45.

H.263 [6] [19] version 2 Interactive and Streaming Wireless Profile (Profile 3) Level 45 should be supported.

ISO/IEC 14496-2 [13] (MPEG-4 Visual) Simple Profile at Level 3 should be supported with the following constraints:

- Number of Visual Objects supported shall be limited to 1.
- The maximum frame rate shall be 30 frames per second.
- The maximum f_code shall be 2.
- The intra_dc_vlc_threshold shall be 0.
- The maximum horizontal luminance pixel resolution shall be 352 pels/line.
- The maximum vertical luminance pixel resolution shall be 288 pels/VOP.
- If AC prediction is used, the following restriction applies: QP value shall not be changed within a VOP (or within a video packet if video packets are used in a VOP). If AC prediction is not used, there are no restrictions to changing QP value.

H.264 (AVC) [41] Baseline Profile at Level 1.1 [42] should be supported with constraint_set1_flag=1 and without requirements on output timing conformance (Annex C of [41]). Each sequence parameter set of H.264 (AVC) shall contain the vui_parameters syntax structure including the num_reorder_frames syntax element set equal to 0.

The H.264 (AVC) decoder in a PSS client shall start decoding immediately when it receives data (even if the stream does not start with an IDR access unit) or alternatively no later than it receives the next IDR access unit or the next recovery point SEI message, whichever is earlier in decoding order. The decoding process for a stream not starting with an IDR access unit shall be the same as for a valid H.264 (AVC) bitstream. However, the client shall be aware that such a stream may contain references to pictures not available in the decoded picture buffer. The display behaviour of the client is out of scope of this specification.

NOTE 1: Terminals may use full-frame freeze and full-frame freeze release SEI messages of H.264 (AVC) to control the display process.

NOTE 2: An H.264 (AVC) encoder should code redundant slices only if it knows that the far-end decoder makes use of this feature (which is signaled with the redundant-pic-cap MIME/SDP parameter as specified in [43]). H.264 (AVC) encoders should also pay attention to the potential implications on end-to-end delay.

NOTE 3: If a codec is supported at a certain level, then all (hierarchically) lower levels shall be supported as well. Examples of lower levels include Level 10 for H.263 Profile 0 and 3, Level 0 for MPEG-4 Visual Simple Profile and Level 1 for H.264 (AVC) Baseline Profile. However, as for instance Level 20 is not hierarchically lower than Level 45 of H.263 Profile 0 and 3, support for Level 45 does not imply support for Level 20.

NOTE 4: All levels are minimum requirements. Higher levels may be supported and used for negotiation.

NOTE 5: If a codec is supported at a certain level, it implies that on the receiving side, the decoder is required to support the decoding of bitstreams up to the maximum capability of this level. On the sending side, the support of a particular level does not imply that the encoder may produce a bitstream up to the maximum capability of the level.

6.3 Real time text

3G PS multimedia terminals offering real time text conversation should support ITU-T Recommendation T.140 [25] Text Conversation presentation coding.

6.4 Interactive and background data

SIP signalling offers initialisation of packet switched interactive or background class reliable data services as well. However specification of such data services are outside the scope of the present document.

6.5 Speech Enabled Service

3G PS multimedia terminals offering speech enabled services should support the DSR Extended Advanced Front-end codec [37]

Speech enabled services may also be supported with AMR or AMR-WB audio codecs, however it is noted that there is a substantial performance advantage from DSR [45].

7 Call control

Functional requirements for call control are specified in 3GPP TS 23.228 [20].

The required signalling functions and call control protocols are specified in 3GPP TS 24.229 [15].

8 Bearer control

The media control is based on declaration of terminal media capability sets in SDP part of appropriate SIP messages.

Relation of application level SDP signalling and radio access bearer assignment is defined outside the present document. The QoS architecture and concept for WCDMA and GERAN is specified in 3GPP TS 23.107 [21]. The end-to-end QoS framework involving GPRS and UMTS is specified in 3GPP TS 23.207 [22]. The applicable general QoS mechanism and service description for the GPRS in GSM and UMTS is specified in 3GPP TS 23.060 [23].

9 Multimedia stream encapsulation

9.1 MIME media types

The terminal shall declare the mandatory and any optional media streams using the codec specific MIME media types in the associated SDP syntax. The MIME media types for the mandatory and optional codecs shall be according to the corresponding types registered by IANA.

- AMR narrowband speech codec MIME media type as specified in annex B.
- AMR wideband speech codec MIME media type is specified in annex B.
- H.263 [6] video codec MIME media type is specified in annex C.
- MPEG-4 visual simple profile level 0 MIME media type as specified in RFC 3016 [5].
- H.264 (AVC) video codec MIME media type is specified in [43].
- ITU-T Recommendation T.140 [25] Text Conversation MIME media type as specified by RFC 4103 [49].
- Telephone-event MIME media type as specified by RFC 2833 [36].
- DSR MIME media type as specified in [38].

9.2 RTP payload

RTP payload formats specified by IETF shall be used for real time media streams.

RTP payload format for the AMR narrowband speech codec is specified in annex B.

RTP payload format for the AMR wideband speech codec is specified in annex B.

RTP payload format for the ITU-T Recommendation H.263 [6] video codec is specified in IETF RFC 2429 [3].

RTP payload format for the MPEG-4 visual simple profile level 0 is specified in IETF RFC 3016 [5].

RTP payload format for the ITU-T Recommendation H.264 (AVC) [41] video codec is specified in [43], where the interleaved packetization mode shall not be used. Receivers shall support both the single NAL unit packetization mode and the non-interleaved packetization mode of [43], and transmitters may use either one of these packetization modes.

RTP payload format for the ITU-T Recommendation T.140 [25] text conversation coding is specified in IETF RFC 4103 [49].

RTP payload format for the telephone-event is specified in IETF RFC 2833 [36].

RTP payload format for the DSR Extended Advanced Front-end is specified in [38].

Annex A (informative): Information on optional enhancements

This annex is intended for informational purposes only. This is not an integral part of the present document.

A.1 Video

This clause gives recommendations for the video codec implementations within 3G PS multimedia terminals.

Regardless of which specific video codec standard is used, all video decoder implementations should include basic error concealment techniques. These techniques may include replacing erroneous parts of the decoded video frame with interpolated picture material from previous decoded frames or from spatially different locations of the erroneous frame. The decoder should aim to prevent the display of substantially corrupted parts of the picture. In any case, it is recommended that the terminal should tolerate *every* possible bitstream without catastrophic behaviour (such as the need for a user-initiated reset of the terminal).

3G PS terminal video encoders and decoders are recommended to support the 1:1 pixel format (square format).

A.1.1 H.263 video codec

H.263 was approved as a standard in 1996. Since then, version 2 and version 3 enhancing version 1 have been approved in 1998 and 2000 respectively. As of today, H.263 contains an extensive set of mandatory and optional coding tools. H.263 [6] annex X [19] defines codec profiles for various target environments.

The Baseline Profile (Profile 0) stands for H.263 with no optional modes of operation. It includes the basic coding tool set common in modern video coding standards. It provides simple means to insert resynchronisation points within the video bitstream, and, therefore, it enables recovery from erroneous or lost data.

The Version 2 Interactive and Streaming Wireless Profile (Profile 3) provides enhanced compression efficiency when compared to the Baseline Profile. Moreover, it provides enhanced error resilience for delivery to wireless devices. Specifically, Profile 3 includes the following optional coding modes:

1) Advanced INTRA Coding (annex I). Use of this mode improves the compression efficiency for INTRA macroblocks (whether within INTRA pictures or predictively-coded pictures);

2) Deblocking Filter (annex J). A deblocking filter improves image quality by reducing blocking artifacts. When compared to deblocking filtering performed as a postprocessing operation, the Deblocking Filter Mode reduces the amount of required memory, as no additional picture memory is needed for the filtered images. This mode also includes the four-motion-vector-per-macroblock feature and picture boundary extrapolation for motion compensation, both of which can further improve compression efficiency;

3) Slice Structured Mode (annex K). This mode provides a flexible mechanism to insert resynchronisation points within the video bitstream for recovery from erroneous or lost data.

4) Modified Quantisation (annex T). This mode enables flexible quantiser control that can be used in sophisticated bit-rate control algorithms. In addition, it improves chrominance fidelity.

[FFS]

A.2 Audio

[FFS]

A.3 Text

Use of the redundancy coding variant specified in RFC 4103 [49] is recommended for error resilience.

Annex B (normative): AMR and AMR-WB RTP payload and MIME type registration

The AMR and AMR-WB speech codec RTP payload, storage format and MIME type registration are specified in [35].

Annex C (normative): ITU-T H.263 MIME media type registration

NOTE: The intention is to replace this normative annex with the IETF RFC defining the H.263 [6] video codec MIME media type registration when the RFC is available.

H.263 video codec MIME media type is specified as follows:

MIME media type name: video;

MIME subtype name: H263-2000;

Required parameters: None;

Optional parameters:

- profile: H.263 profile number, in the range 0 through 8, specifying the supported H.263 annexes/subparts;
- level: Level of bitstream operation, in the range 0 through 99, specifying the level of computational complexity of the decoding process. When profile and level parameters are not specified, Baseline Profile (Profile 0) Level 10 are the default values.

The profile and level specifications can be found in [19]. Note that the RTP payload format for H263-2000 is the same as for H263-1998 published in RFC 2429 [3], but additional annexes/subparts are specified along with the profiles and levels.

Annex D (informative): Push-to-Talk over cellular (PoC)

For PoC the audio codecs specified in section 6.1, namely AMR or AMR-WB are applicable. Speech codec bit rates and transport formats settings have to be selected considering the available transmission bandwidth and the allowable transport delay. In order not to introduce undue delay for RTP packetization, it is recommended to limit the number of speech codec frames per packet to 20 and not to use interleaving.

Under the assumption of RTP packetization according to [35] using octet-aligned mode, no interleaving and using 10 frames per RTP packet and depending on the IP version in IMS, the following tables show the required bandwidth for the available AMR and AMR-WB speech codec modes. Bandwidth restrictions may imply that only the lowest AMR/AMR-WB modes can be used for PoC. In order to maximize speech quality, it is recommended to use the respective highest possible bit rate.

Table 1: Required bandwidth for PoC using AMR

AMR Mode	Required bandwidth when IPv4 is used [bits/s] [Note]	Required bandwidth when IPv6 is used [bits/s]
AMR 4.75	6840	7640
AMR 5.15	7240	8040
AMR 5.9	8040	8840
AMR 6.7	8840	9640
AMR 7.4	9640	10440
AMR 7.95	10040	10840
AMR 10.2	12440	13240
AMR 12.2	14440	15240
Note: For the usage of IP version in IMS see TS 23.221 [44], subclause 5.1.		

Table 2: Required bandwith for PoC using AMR-WB

AMR-WB Mode	Required bandwidth when IPv4 is used [bits/s] [Note]	Required bandwidth when IPv6 is used [bits/s]
AMR-WB 6.60	8840	9640
AMR-WB 8.85	11240	12040
AMR-WB 12.65	14840	15640
AMR-WB 14.25	16440	17240
AMR-WB 15.85	18040	18840
AMR-WB 18.25	20440	21240
AMR-WB 19.85	22040	22840
AMR-WB 23.05	25240	26040
AMR-WB 23.85	26040	26840
Note: For the usage of IP version in IMS see TS 23.221 [44], subclause 5.1.		

Annex E (informative): Change history

Change history							
Date	**TSG SA#**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
03-2001	11	SP-010095			Version for Release 4		4.0.0
06-2001	12	SP-010309	001		Update of AMR-NB and AMR-WB RTP payload	4.0.0	4.1.0
	12				Release 4 version withdrawn		
06-2001	12	SP-010378	002		Applicability of TS 26.235 to GERAN FFS	4.0.0	5.0.0
	12				TS 26.235 approved at TSG-SA#12 for Release 5		5.0.0
03-2002	15	SP-020089	003	2	Update of AMR & AMR-WB RTP payload format	5.0.0	5.1.0
03-2002	15	SP-020154	004		Corrections of references to obsolete SIP RFC 2543 IETF specification	5.0.0	5.1.0
06-2003	20	SP-030218	005	1	Handling of DTMF in IMS	5.1.0	6.0.0
06-2004	24	SP-040356	006	4	Introduction of the DSR codec	6.0.0	6.1.0
09-2004	25	SP-040653	007	1	Language improvement and alignment	6.1.0	6.2.0
09-2004	25	SP-040658	008	1	Introduction of the H.264 video codec into packet-switched conversational services	6.1.0	6.2.0
09-2004	25	SP-040653	009		Support for 128 kbps video in the packet-switched conversational services	6.1.0	6.2.0
12-2004	26	SP-040843	010	1	Inclusion of PoC support	6.2.0	6.3.0
12-2004	26	SP-040837	011		Add reference to TR 26.943	6.2.0	6.3.0
03-2005	27	SP-050097	012	1	Correction of inconsistency regarding the maximum number of speech frames per RTP packets for PoC	6.3.0	6.4.0
03-2005	27	SP-050099	013	1	Correction of reference	6.3.0	6.4.0
12-2005	30	SP-050790	0014	3	CSI interoperability of media types and formats	6.4.0	7.0.0
03-2006	31	SP-060219	0015	3	Update of a RFC on real time text	7.0.0	7.1.0
09-2007	37	SP-070627	0017		Correction of references	7.1.0	7.2.0
12-2007	38	SP-070760	0018	1	Correction of video codecs for PS conversational multimedia applications	7.2.0	7.3.0
03-2008	39	SP-080008	0019	1	Scope clarifications and editorial changes	7.3.0	7.4.0
12-2008	42				Version for Release 8	7.4.0	8.0.0
12-2009	46				Version for Release 9	8.0.0	9.0.0
03-2011	51				Version for Release 10	9.0.0	10.0.0
09-2012	57				Version for Release 11	10.0.0	11.0.0