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Technical Specification

**Digital cellular telecommunications system (Phase 2+);
Universal Mobile Telecommunications System (UMTS);
LTE;
Application of SIP-I Protocols to Circuit Switched (CS) core network architecture;
Stage 3
(3GPP TS 29.231 version 8.1.0 Release 8)**

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Foreword

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Contents

Foreword

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

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:

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1 Scope

The present document describes the protocols to be used when SIP-I is optionally used as call control protocol in a 3GPP CS core network on Nc interface, see 3GPP TS 23.231 [1]. The SIP-I protocol operates between (G)MSC servers. The SIP-I architecture consists of a number of protocols. The following types of protocols are described: call control protocol, resource control protocols and user plane protocol for this architecture. The architecture complies with the requirements imposed by 3GPP TS 23.231 [1] and TS 23.153 [2].

Interworking of SIP-I on Nc to external networks is described by TS 29.235 [3].

The present document is valid for a 3rd generation PLMN (UMTS) complying with Release 8 and later.

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] 3GPP TS 23.231: "SIP-I based Circuit Switched Core Network ; Stage 2"

[2] 3GPP TS 23.153: "Out of Band Transcoder Control; Stage 2"

[3] 3GPP TS 29.235: "Interworking between the 3GPP CS domain with SIP-I as signalling protocol and other networks"

[4] ITU-T Recommendation Q.1912.5: "Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part".

[5] IETF RFC 2046 (November 1996): "Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types".

[6] IETF RFC 3966: "The tel URI for Telephone Numbers".

[7] IETF RFC 2976: "The SIP INFO method".

[8] IETF RFC 3204: "MIME media types for ISUP and QSIG Objects".

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

[10] IETF RFC 3262: "Reliability of Provisional Responses in the Session Initiation Protocol (SIP)".

[11] IETF RFC 3264: "An Offer/Answer Model with the Session Description Protocol (SDP)".

[12] IETF RFC 3311: "The Session Initiation Protocol (SIP) UPDATE Method".

[13] IETF RFC 3312: "Integration of Resource Management and Session Initiation Protocol (SIP)".

[14] IETF RFC 3323: "A Privacy Mechanism for the Session Initiation Protocol (SIP)".

[15] IETF RFC 3325: "Private Extensions to the Session Initiation Protocol (SIP) for Network Asserted Identity within Trusted Networks".

[16] IETF RFC 3326: "The Reason Header Field for the Session Initiation Protocol (SIP)".

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

[18] 3GPP TS 29.232: "Media Gateway Controller (MGC) - Media Gateway (MGW) interface; Stage 3".

[19] 3GPP TS 29.415: "Core network Nb data transport and transport signalling".

[20] 3GPP TS 29.414: "Core Network Nb data transport and transport signalling".

[21] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".

[22] IETF RFC 3389 (September 2002): "Real-time Transport Protocol (RTP) Payload for Comfort Noise".

[23] IETF RFC 4733 "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals".

[24] IETF RFC 4028 "Session Timers in the Session Initiation Protocol (SIP)".

[25] IETF RFC 2960: "Stream Control Transmission Protocol".

[26] IETF RFC 3309: "Stream Control Transmission Protocol (SCTP) Checksum Change"

[27] IETF RFC 4168: "The Stream Control Transmission Protocol (SCTP) as a Transport for the Session Initiation Protocol (SIP)".

[28] IETF draft "Example calls flows of race conditions in the Session Initiation Protocol (SIP) draft-ietf-sipping-race-examples-05"

Editor's Note: this draft should be changed to a full RFC if available or removed if IETF RFC 3261 is replaced and the associated issues resolved.

[29] IETF RFC 791: "Internet Protocol (IP)"

[30] IETF RFC 2460: "Internet Protocol, Version 6 (IPv6)"

[31] IETF RFC 3550: "RTP: A Transport Protocol for Real-Time Applications".

[31] 3GPP TS 26.102 " Mandatory speech codec; Adaptive Multi-Rate (AMR) speech codec;Interface to Iu, Uu and Nb".

[32] 3GPP TS 26.103 "Speech codec list for GSM and UMTS".

3 Definitions, symbols and abbreviations

3.1 Definitions

3.2 Symbols

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

Nc Interface between the(G)MSC servers.
Mc Interface between the server and the media gateway.
Nb Interface between media gateways (MGW).

3.3 Abbreviations

For the purposes of the present document, the abbreviations given in 3GPP TR 21.905 [21] and the following apply. An abbreviation defined in the present document takes precedence over the definition of the same abbreviation, if any, in 3GPP TR 21.905 [21].

BCU Bearer Control Unit

BICC Bearer Independent Call Control
MGC Media Gateway Controller
MIME Multi-purpose Internet Mail Extensions
OoBTC Out of Band Transcoder Control

4 Protocols

4.1 Introduction

Implementations providing any of the interfaces or protocols identified in the subclauses below shall implement the requirements of the specifications identified in those subclauses.

4.2 Call control protocol (Nc interface)

Table 4.2.1: Call Control Protocol Specifications

Identification	Protocol Name	Amendments and Endorsements to referenced specifications	Support
Q.1912.5 [4]	Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part	See Clause 5.1	Mandatory
IETF RFC 2046 [5]	Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types	See Clause 5.2	Mandatory
IETF RFC 3966 [6]	URLs for Telephone Calls	See Clause 5.4	Mandatory
IETF RFC 2976 [7]	The SIP INFO Method	See Clause 5.5	Mandatory
IETF RFC 3204 [8]	MIME media types for ISUP and QSIG Objects	See Clause 5.6	Mandatory
IETF RFC 3261 [9]	SIP: Session Initiation Protocol	See Clause 5.7	Mandatory
IETF RFC 3262 [10]	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)	See Clause 5.8	Mandatory
RFC 3264 [11]	An Offer/Answer Model with the Session Description Protocol (SDP)	See Clause 5.9	Mandatory
IETF RFC 3311 [12]	The Session Initiation Protocol UPDATE Method	See Clause 5.10	Mandatory
IETF RFC 3312 [13]	Integration of Resource Management and Session Initiation Protocol (SIP)	See Clause 5.11	Mandatory
IETF RFC 3323 [14]	A Privacy Mechanism for the Session Initiation Protocol (SIP)	See Clause 5.12	Mandatory
IETF RFC 3325 [15]	Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks	See Clause 5.13	Mandatory
IETF RFC 3326 [16]	The Reason Header Field for the Session Initiation Protocol (SIP)	See Clause 5.14	Mandatory
IETF RFC 4566 [17]	SDP: Session Description Protocol	See Clause 5.3	Mandatory
IETF RFC 4028 [24]	Session Timers in the Session Initiation Protocol (SIP)	See Clause 5.15	Optional

4.3 Resource control protocol (G)MSC and MGW (Mc Interface)

Table 4.3.1: Resource Control Protocol Specifications

Identification	Protocol Name	Amendments and Endorsements to referenced specifications	Support
3GPP TS 29.232. [18]	Media Gateway Controller (MGC) – Media Gateway (MGW) Interface; Stage 3	None (NOTE)	Mandatory
NOTE: IPv4 (IETF RFC 791 [29]) shall be supported on the Mc interface. IPv6 (IETF RFC 2460 [30]) may be supported.			

4.4 Bearer Framing Protocol between MGWs (Nb interface)

Table 4.4.1: Framing Protocol Specifications

Identification	Protocol Name	Amendments and Endorsements to referenced specifications	Support
IETF RFC 3550[31]	RTP: A Transport Protocol for Real-Time Applications	None	Mandatory
NOTE: Further specified by 3GPP TS 29.414 [20]			

4.5 Signalling Transport

4.5.1 Call Control protocols

Table 4.5.1.1: Call Control Signalling Transport

Identification	Protocol Name	Amendments and Endorsements to referenced specifications	Support
IETF RFC 791 [29]	Internet Protocol, Version 4 (IPv4)	See clause 5.19	Mandatory
IETF RFC 2460 [30]	Internet Protocol, Version 6 (IPv6)"	See clause 5.20	Optional
IETF RFC 2960 [25]	Stream Control Transmission Protocol	See clause 5.16	Mandatory
IETF RFC 3309 [26]	Stream Control Transmission Protocol (SCTP) Checksum Change	See clause 5.17	Mandatory
IETF RFC 4168 [27]	The Stream Control Transmission Protocol (SCTP) as a Transport for the Session Initiation Protocol (SIP)	See clause 5.18	Mandatory

4.5.2 Resource control protocol (G)MSC and MGW (Mc Interface)

Table 4.5.2.1: Resource Control Signalling Transport

Identification	Protocol Name	Amendments and Endorsements to referenced specifications	Support
3GPP TS 29.232 [18]	Media Gateway Controller (MGC) – Media Gateway (MGW) Interface;Stage 3	None	Mandatory

4.5.3 IP Transport between MGWs (Nb interface)

Table 4.5.3.1: Nb Interface Signalling Transport

Identification	Protocol Name	Amendments and Endorsements to referenced specifications	Support
3GGP TS 29.414 [20]	Core Network Nb Data Transport and Transport Signalling	None	Mandatory

4.6 Payload Types

The details of which payload types shall be supported for the SIP-I application are defined in 3GPP TS 26.102 [31] and the RTP attributes for each specific codecs are defined in 3GPP TS 26.103 [32].

5 Amendments and Endorsements to Referenced Specifications

5.1 ITU-T Q.1912.5 (Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part)

Only Profile C shall apply.

5.2 IETF RFC 2046 (Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types)

The "multipart" MIME type with the sub-type of "mixed" shall be supported as per IETF RFC 2046 [5] to allow exchange of multiple bodies in a single SIP message (e.g. initial INVITE message with ISUP IAM encapsulation and SDP bodies) between (G)MSC-Servers. Nested MIME message is not supported in the SIP-I based Nc interface.

The following MIME types only shall be supported in the SIP-I based Nc interface:

- The "application" MIME type as per IETF RFC 2046 [5] with the sub-type of "ISUP" as per IETF RFC 3204 [8] to allow exchange of ISUP encapsulation in SIP messages between (G)MSC-Servers.
- The "application" MIME type as per IETF RFC 2046 [5] with the sub-type of "SDP" as per IETF RFC 4566 [17] to allow exchange of SDP in SIP messages between (G)MSC-Servers.

5.3 IETF RFC 4566 (SDP: Session Description Protocol)

The following SDP properties are described for SIP-I call control procedures; use of SDP within H.248 MGW control protocol interface is described in the relevant profile specification, e.g. 3GPP TS 29.232 [18].

Table 5.3.1: Support of SDP Fields

field	Meaning	Support	Comments
v	Protocol Version	Mandatory	The value shall be 'v=0'
o	Origin	Mandatory	
s	Session Name	Mandatory	
c	Connection Data	Mandatory	
t	Timing	Mandatory	The value shall be 't=0 0'
a	Attributes	Conditional	See table 5.3.2.
m	Media Descriptions	Mandatory	
b=RS, b=RR	RTCP bandwidth modifiers	Conditional	A (G)MSC wishing to deactivate RTCP or using RTCP only to negotiate the use of RTP bearer multiplexing and RTP header compression (see 3GPP TS 29.414 [20]) shall set the RTCP bandwidth modifiers to zero. If a (G)MSC receives SDP bandwidth modifiers for RTCP equal to zero, it should reply by setting its RTCP bandwidth using SDP bandwidth modifiers with values equal to zero.
NOTE: Fields not listed in the table may be ignored by the (G)MSC Server on receipt of the SDP.			

Table 5.3.2: Support of Attributes

attr	Meaning	Support	Comments
rtpmap	RTP map	Conditional	Mandatory for dynamic payload type and optional for static payload type.
recvonly	Receive only	Mandatory	
sendrecv	Send and receive	Mandatory	
sendonly	Send only	Mandatory	
inactive	Inactive	Mandatory	
fmtp	format specific parameters	Conditional	It is dependent upon the payload type if format specific parameters are required or not.
maxptime	Maximum packet size in ms	See 3GPP TS 26.102 [31]	
ptime	Packetisation length	See 3GPP TS 26.102 [31]	
NOTE: Attributes not listed in the table may be ignored by the (G)MSC Server on receipt of the SDP.			

5.4 IETF RFC 3966 (The tel URI for Telephone Numbers)

5.5 IETF RFC 2976 (The SIP INFO method)

The SIP INFO method shall be supported as per IETF RFC 2976 [7] to allow exchange of ISUP signalling between (G)MSC-Servers. The contents of the message body shall not be encrypted. The SIP INFO method shall not be used to carry any other kind of information, e.g. DTMF digits.

5.6 IETF RFC 3204 (MIME media types for ISUP and QSIG Objects)

Only ISUP Media Type is supported in the SIP-I based Nc interface, see ITU-T Q.1912.5 [4] Clause 5.4.1.2.

The "version" parameter is used to signal the ISUP version, as per ITU-T Q.1912.5 [4], sub-clause 5.4.1.2.

5.7 IETF RFC 3261(SIP: Session Initiation Protocol)

SIP-I initial and any subsequent INVITEs shall always include SDP.

3GPP SIP-I entities shall apply loose routeing on SIP-I based Nc.

SIP forking is not supported in the SIP-I Nc interface.

Race conditions for call clearing should be solved as described in clause 3.1.2 of the IETF draft "Example calls flows of race conditions in the Session Initiation Protocol (SIP) draft-ietf-sipping-race-examples-05" [27].

5.8 IETF RFC 3262 (Reliability of Provisional Responses in the Session Initiation Protocol)

IETF RFC 3262 [10] specifies an extension to SIP in order to provide reliable provisional response messages. As support for PRACK's is required for a SIP-I based Nc interface to support preconditions, the support of 100rel is mandatory for the 3GPP SIP-I profile on the Nc interface.

Procedures in clauses 3 and 4 of IETF RFC 3262 [10] apply on the Nc interface according to the following rules:

- A (G)MSC Server originating a SIP INVITE shall advertise its preference of provisional reliable responses via a SUPPORTED header containing the tag "100Rel".

- A (G)MSC Server receiving a SIP INVITE will receive a SUPPORTED header containing the tag "100rel" and shall include a REQUIRE header with tag "100rel" and RSeq header field when sending a response in the range 101-199.

- A (G)MSC Server receiving a response in the range 101-199 with a REQUIRE header present with tag "100rel" shall generate a PRACK request for this provisional response.

Authentication procedures are not required to be supported for PRACK (and any other) messages.

5.9 IETF RFC 3264 (An Offer/Answer Model with the Session Description Protocol)

5.9.1 Multicast Streams

Procedures in Clauses 5.2 and 6.2 of IETF RFC 3264 do not need to be supported.

5.9.2 3GPP Node Generating the Offer

Procedures in Clause 5.1 of IETF RFC 3264 apply with the following modifications:

- An MSC-S initiating an offer with multiple speech codec payload types in one m-line shall apply the related procedures in Clause 9 of 3GPP TS 23.153 [2], in particular the following requirement from IETF RFC 3264 Clause 5.1 is overruled:

  "*Once the offerer has sent the offer, it MUST be prepared to receive media for any recvonly streams described by that offer. It MUST be prepared to send and receive media for any sendrecv streams in the offer, and send media for any sendonly streams in the offer (ofcourse, it cannot actually send until the peer provides an answerwith the needed address and port information).*"

5.9.3 3GPP Node Generating the Answer

Procedures in Clause 6 of IETF RFC 3264 apply with the following modifications:

- If the 3GPP MSC-S terminating the codec negotiation supports multiple speech codec payload types it shall apply the related procedures in Clause 9 of 3GPP TS 23.153 [2]. It does not need to be prepared to receive media encoded by speech codec payload types within the answered Available Codec List (see Clause 6.1.1).

5.9.4 3GPP Node as Offerer Processing of the Answer

Procedures in Clause 7 of IETF RFC 3264 apply with the following modifications:

- If the offering MSC supports multiple speech codecs and the MSC-S receives an answer with multiple speech codec payload types in one m-line, it shall apply the related procedures in Clause 9 of 3GPP TS 23.153 [2]. It does not need to be prepared to receive media encoded by speech codec payload types within the Available Codec List (see Clause 6.1.1) received in the answer.

5.9.5 Modifying the session

Procedures in Clause 8 of IETF RFC 3264 apply with the same modifications as described in Clauses 5.9.2, 5.9.3, and 5.9.4.

5.9.6 Unspecified Connection Address

The use of an "unspecified connection address" may be used during initial call establishment, for deferred MGW selection where no user plane connection has been seized by the Offerer.

No further applications for the use of an "unspecified connection address" are defined in SIP-I on Nc.

The "unspecified connection address" shall not be sent within a re-INVITE message.

For IPv6 implementations the "unspecified connection address" shall be defined by a domain name within the ".invalid" DNS top-level domain.

For IPv4 implementations if the "unspecified connection address" shall be defined by the IPv4 unspecified address (c= IN IP4 0.0.0.0).

5.10 IETF RFC 3311 (The Session Initiation Protocol (SIP) UPDATE Method)

The SIP UPDATE method shall be supported to allow updating of session parameters (media streams, codecs) during early and confirmed dialog, and allow the support of preconditions.

The support of the UPDATE method shall be negotiated on SIP-I based Nc according to the following rules:

- A (G)MSC Server originating a SIP INVITE request shall advertise its support of the UPDATE method via the ALLOW header listing the UPDATE method.
- A (G)MSC Server receiving a SIP INVITE request shall include an ALLOW header listing the UPDATE method when sending a response in the range 101-199. The MSC-S is then allowed to generate the UPDATE method, for the purpose of session modification during early dialog. In addition the MSC-S shall include an ALLOW header listing the UPDATE method when sending a 2xx final response.
- A (G)MSC Server receiving a response to a SIP INVITE request with an ALLOW header present listing the UPDATE method is then allowed to generate the UPDATE method.

Authentication and Integrity protection procedures are not required to be supported for the UPDATE message.

5.11 IETF RFC 3312 (Integration of Resource Management and Session Initiation Protocol)

Precondition type QoS only shall be supported using the segmented status type (therefore end-to-end status type e.g. Clause 13.1 does not apply) and the strength-tag value "mandatory" for the local segment and the strength-tag value "optional" for the remote segment. Precondition tag may be included in either REQUIRE or SUPPORTED header; the use of the SUPPORTED header is a deviation from this RFC when the strength-tag contains a "mandatory" value.

5.12 IETF RFC 3323 (A Privacy Mechanism for the Session Initiation Protocol)

5.13 IETF RFC 3325 (Private Extensions to the Session Initiation Protocol (SIP) for Network Asserted Identity within Trusted Networks)

5.14 IETF RFC 3326 (The Reason Header Field for the Session Initiation Protocol)

The SIP Reason header provides a means of transporting SIP or ISUP/BICC cause value in the SIP message; this is especially useful in SIP message without encapsulated ISUP information.

The SIP Reason header shall be supported as specified by IETF RFC 3326 [16] and by ITU-T Q.1912.5 [4] (as endorsed by sub-clause 5.1 of the present specification), with the following modifications:

- a Reason header field containing a Q.850 cause value shall be added to the SIP CANCEL message sent by the (G)MSC Server if the value is known; if unknown, the Q.850 cause value 31 (normal unspecified) shall be sent.
- the header field shall not be protected by any integrity mechanism.

5.15 IETF RFC 4028 (Session Timers in the Session Initiation Protocol)

SIP session continuity may be supported through the use of SIP Session Timer as per IETF RFC 4028 [24] with the following amendments.

- Clause 4 indicates that the minimum value of the *Session-Expires* header field should not be less than 30 minutes. Since the vast majority of mobile calls are significantly less than 30 minutes, it may be desirable to use a minimum *Session-Expires* of less than 30 minutes within the 3GPP CS core network. The RFC states that the value SHOULD NOT be less then 30 minutes. That statement is modified to "…MAY choose values of less than 30 minutes but greater than the absolute minimum of 90 seconds."

- Clause 8 "Proxy Behavior" is not applicable to MSC Servers.

5.16 IETF RFC 2960 (Stream Control Transmission Protocol)

See sub-clause 5.18.

5.17 IETF RFC 3309 (Stream Control Transmission Protocol (SCTP) Checksum Change)

See sub-clause 5.18.

5.18 IETF RFC 4168 (The Stream Control Transmission Protocol (SCTP) as a Transport for the Session Initiation Protocol)

SCTP shall be the transport protocol for a SIP-I based Nc interface as per IETF RFC 4168 [27].

Semi-permanent SCTP associations shall be established between peer 3GPP SIP-I entities, i.e. the SCTP associations shall remain up under normal circumstances.

Local multi-homing should be supported. Remote multi-homing shall be supported.

Multiple local SCTP endpoints may be supported. Multiple remote SCTP endpoints shall be supported. When multiple local or remote SCTP endpoints are configured, several simultaneous SCTP associations shall be supported between peer 3GPP SIP-I entities.

Published IP addresses and ports for SCTP associations are supported as specified below:

- The (G)MSC Server shall support SCTP association setup, where an SCTP association request is performed to a published IP address and port.

- The (G)MSC Server may initiate SCTP association requests from a local published IP address and port.

- The (G)MSC Server shall accept incoming SCTP association requests from remote published IP address and port.

- When two endpoints are configured for a specific SCTP association between two published IP addresses and ports, one endpoint shall be configured as "server" and the peer endpoint as the "client" with respect to SCTP association setup.

NOTE: Published IP addresses or ports can be configured via O&M or they can be obtained through a DNS enquiry.

Dynamically assigned ports for SCTP associations are supported as specified below:

- The (G)MSC Server may initiate a SCTP association from an ephemeral (non-published) port.

- The (G)MSC Server shall accept incoming SCTP association requests from remote ephemeral (non-published) ports.

Checksum calculation for SCTP shall be supported as specified in RFC 3309 [26] instead of the method specified in IETF RFC 2960 [25].

5.19 IETF RFC 791 (Internet Protocol, Version 4)

IPv4 (IETF RFC 791 [29]) shall be supported on the Nc interface.

5.20 IETF RFC 2460 (Internet Protocol, Version 6)

IPv6 (IETF RFC 2460 [30]) may be supported on the Nc interface.

6 3GPP Extensions

6.1 Codec Negotiation

6.1.1 Encoding of 3GPP_OoBTC_Indicator

3GPP OoBTC Indicator shall be encoded as the following media-level SDP attribute with the following syntax (ABNF definition):

3GPP_OoBTC_Indicator = "a=3gOoBTC"

No attribute values to this SDP attribute are defined in the present Release. If attribute values for the attribute are received, they shall be ignored.

The SDP offer and SDP answer signalling procedures for OoBTC Indicator are described in Clause 9 of 3GPP TS 23.153 [2].

Editor´s Note: The 3GPP_OoBTC_Indicator will be registered at IANA.

6.1.2 Encoding of SDP answer including 3GPP OoBTC Indicator

If the 3GPP OoBTC Indicator is included in an SDP answer, the corresponding SDP m-line shall be encoded as follows:

- The first codec in the m-line (indicated by RTP payload type) shall indicate the Selected Codec.
- Any subsequent codecs in the m-line (indicated by RTP payload type), which are not of "auxiliary" payload type (see next bullet), shall indicate the Available Codec List (ACL)
- Codecs of "auxiliary" payload type, i.e. RTP Telephony Event payload type (IETF RFC 4733 [23]) or the comfort noise codec (IETF RFC 3389 [22]), may be included as the last codecs in the m-line (indicated by RTP payload type).

6.2 MGW Identifier

6.2.1 Semantic and Usage of the MGW_Identifier

The MGW_Identifier shall be used to encode a MGW Identity as used for the optional "optimised MGW selection " and "deferred MGW selection" procedures in Clauses 4.4.2 and 4.4.3 of TS 23.231 [3].

The support of the MGW_Identifier attribute is only required for a (G)MSC Server that supports at least one of the optional "optimised MGW selection " and "deferred MGW selection" procedures.

6.2.2 Encoding of MGW_Identifier

The MGW Identifier shall be encoded as the following "session-level" value attribute with the following syntax (ABNF definition):

MGW_Identifier = "a=MGW_Identifier: <MGW_Id>"

<MGW_Id> = *Octet string containing any octet value except 0x00 (Nul), 0x0A (LF), and 0x0D (CR). Values are to be interpreted as in ISO-10646 character set with UTF-8 encoding.*

NOTE: The <MGW_Id> sub-field may be encoded for example in the same manner as BCU-ID in BICC, i.e. 4 Octets for representing Network ID field and Local BCU-ID field.

<MGW_Id> shall contain an operator-defined unique identifier for a MGW.

Attribute values of the SDP MGW_Identifier attribute shall not be subject to the SDP "charset" attribute.

Editor´s Note: The MGW_Identifier will be registered at IANA.

6.2.3 Procedures related to MGW_Identifier

Provided that the (G)MSC Server supports the MGW Identifier attribute, it shall apply the following procedures:

- For the optimised MGW selection, the (G)MSC Server shall apply the MGW Identity related procedures in Clause 4.4.2 of TS 23.231 [3].
- For the deferred MGW selection, the (G)MSC Server shall apply the MGW Identity related procedures in Clause 4.4.3 of TS 23.231 [3].

Annex A (informative): IANA Registration of OoBTC Indicator

A.1 Introduction

This Annex describes information required for the IANA registration of the OoBTC Indicator SDP attribute, as required according to Clause 8.2.4 of IETF RFC 4566 [17]

A.2 Contact name, email address, and telephone number

3GPP Specifications Manager

3gppContact@etsi.org

+33 (0)492944200

A.3 Attribute Name (as it will appear in SDP)

3gOoBTC

A.4 Long-form Attribute Name in English

3GPP Out-of-band Transcoder Control Indicator

A.5 Type of Attribute

Media level

A.6 Is Attribute Value subject to the Charset Attribute?

Not applicable, as no attribute values are defined.

A.7 Purpose of the attribute

The semantics of the 3GPP OoBTC Indicator are defined in Clause 9.4 of 3GPP TS 23.153 [2].

A.8 Appropriate Attribute Values for this Attribute

No attribute values are defined.

Annex B (informative): IANA Registration of MGW Identifier

B.1 Introduction

This Annex describes information required for the IANA registration of the MGW Identifier SDP attribute, as required according to Clause 8.2.4 of IETF RFC 4566 [17]

B.2 Contact name, email address, and telephone number

3GPP Specifications Manager

3gppContact@etsi.org

+33 (0)492944200

B.3 Attribute Name (as it will appear in SDP)

MGW_Identifier

B.4 Long-form Attribute Name in English

Media GateWay Identifier

B.5 Type of Attribute

Session level

B.6 Is Attribute Value subject to the Charset Attribute?

No

B.7 Purpose of the attribute

As defined in Clause 6.2.1

B.8 Appropriate Attribute Values for this Attribute

As defined in Clauses 6.2.2 and 6.2.3

Annex C (informative): Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
CT#36					Agreed as base version for further work		0.0.0
10/2007					Output from CT4#36bis	0.0.0	0.1.0
11/2007					Output from CT4#37	0.1.0	0.2.0
11/2007					Updated with missing contribution from Kobe	0.2.0	0.2.1
02/2008					Output from CT4#38	0.2.1	0.3.0
03/2008	TSG #39				Agreed as information	0.3.0	1.0.0
04/2008					Output from CT4#38Bis	1.0.0	1.1.0
05/2008					Output from CT4#39	1.1.0	1.2.0
07/2008					Output from CT4#39bis	1.2.0	1.3.0
08/2008					Output from CT4#40	1.3.0	1.4.0
09/2008	TSG#41	CP-080484			V2.0.0 was presented for approval	1.4.0	2.0.0
09/2008	TSG#41	CP-080484			V2.0.0 was approved in CT#41	2.0.0	8.0.0
12/2008	TSG#42	CP-080686	0002		Correction on definition of MGW identifier	8.0.0	8.1.0
		CP-080686	0004	1	Clean-up of tables, text and references		
		CP-080686	0005	2	SCTP Usage for SIP-I on Nc		
		CP-080686	0006	1	Format of the unspecified connection address		
		CP-080686	0007	2	Non-Support of SIP forking in SIP-I CSCN		
		CP-080686	0011		Removal of Re-INVITE without SDP		

History

Document history		
V8.1.0	January 2009	Publication