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Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Interworking between the 3GPP CS domain with BICC or ISUP as signalling protocol and external SIP-I networks (3GPP TS 29.164 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).

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1 Scope

The present document defines interworking procedures between a 3GPP CS domain (see 3GPP TS 23.205 [2]) which applies either BICC or ISUP as signalling protocol, and external networks that use SIP-I (see ITU-T Q.1912.5 [20], Profile C) as signalling protocol. The document also describes the related interworking architecture. The control plane interworking is performed by an interworking unit at the interconnection between the 3GPP CS domain and an external SIP-I network. The user plane interworking is performed by an MGW. The present document defines stage 2 procedures for the control of the MGW.

The present specification reuses existing interworking procedures of 3GPP TS 29.163 [4] and ITU-T Q.1912.5 [20], Profile C, as far as possible.

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 TR 21.905: "Vocabulary for 3GPP Specifications".

[2] 3GPP TS 23.205: "Bearer-independent circuit-switched core network; Stage 2".

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

[4] 3GPP TS 29.163: "Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks"

[5] 3GPP TS 29.205: "Application of Q.1900 series to Bearer Independent CS Network architecture; Stage 3".

[6] 3GPP TS 29.332: "Media Gateway Control Function (MGCF) – IM-Media Gateway (IM-MGW) interface, Stage 3".

[7] 3GPP TS 29.414: "Core network Nb data transport and transport signalling".

[8] 3GPP TS 29.415: "Core network Nb interface user plane protocols".

[9] ITU-T Recommendation E.164 (05/1997): "The international public telecommunication numbering plan".

[10] ITU-T Recommendation G.711: "Pulse Code Modulation (PCM) of voice frequencies".

[11] ITU-T Recommendation H.248.1 (2002): "Gateway control protocol: Version 2".

[12] ITU-T Recommendation I.363.5 (1996): "B-ISDN ATM Adaptation Layer specification: Type 5 AAL".

[13] ITU-T Recommendation Q.701 to Q.709: "Functional description of the message transfer part (MTP) of Signalling System No. 7".

[14] ITU-T Recommendation Q.730 to Q.737 (12/1999): "ISDN user part supplementary services".

[15] ITU-T Recommendations Q.761to Q.764 (2000): " Signalling System No.7 ISDN User Part".

[16]   ITU-T Recommendation Q.765 (2000): "Signalling System No. 7 – Application transport mechanism"

[17]   ITU-T Recommendation Q.765.5 (2000): "Signalling system No. 7 – Application transport mechanism: Bearer Independent Call Control (BICC)"

[18]   ITU-T Recommendation Q.850 (1998): "Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part".

[19]   ITU-T Recommendations Q.1902.1 to Q.1902.6 (07/2001): "Bearer Independent Call Control".

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

[21]   ITU-T Recommendation Q.2110 (1994): "B-ISDN ATM adaptation layer - Service Specific Connection Oriented Protocol (SSCOP)".

[22]   ITU-T Recommendation Q.2140 (1995): "B-ISDN ATM adaptation layer - Service specific coordination function for signalling at the network node interface (SSCF AT NNI)".

[23]   ITU-T Recommendation Q.2210 (1996): "Message transfer part level 3 functions and messages using the services of ITU-T Recommendation Q.2140".

[24]   IETF RFC 791: "Internet Protocol".

[25]   IETF RFC 768: "User Datagram Protocol".

[26]   IETF RFC 793: "Transmission Control Protocol".

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

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

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

[30]   IETF RFC 2976: "The SIP INFO Method"

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

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

[33]   IETF RFC 3262: "Reliability of provisional responses".

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

[40]   IETF RFC 3311: "SIP UPDATE method".

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

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

[37]   3GPP TS 29.202: "Core Signalling System No. 7 (SS7) signalling transport in core network; Stage 3".

[38]   3GPP TS 29.231: "Application of SIP-I Protocols to Circuit Switched (CS) core network architecture; Stage 3".

[39]   3GPP TS 29.235: "Interworking between the 3GPP CS domain with SIP-I as signalling protocol and other networks".

3   Definitions and abbreviations

3.1   Definitions

For the purposes of the present document, the terms and definitions given in 3GPP TR 21.905 [1], in
3GPP TS 29.163 [4], in ITU-T E.164 [9], in ITU-T Q.1912.5 [20] and the following apply. A term defined in the
present document takes precedence over the definition of the same term, if any, in any other of the specifications listed
above.

Interworking Unit (IWU): Logical entity that interworks SIP-I signalling with BICC or ISUP signalling in the 3GPP
CS Domain.

Incoming Interworking Unit (I-IWU): Logical entity that terminates incoming calls from the external SIP-I network
side and originates outgoing calls towards the CS Domain side using the BICC or ISUP protocols.

Outgoing Interworking Unit (O-IWU): Logical entity that terminates incoming BICC or ISUP calls from the CS
Domain side and originates outgoing calls towards external SIP-I network.

External SIP-I signalling function: function in the external network, which has the capabilities to process SIP with
encapsulated ISUP messages.

For references to 3GPP TS 29.163 [4] procedures within the present specification, the MGCF in 3GPP TS 29.163 [4] is
to be understood as IWU. The IM-MGW is to be understood as MGW. The Mn interface is to be understood as
interface between IWU and MGW. The IM CN subsystem is to be understood as external SIP-I network. The CS
network is to be understood as 3GPP CS domain.

3.2   Abbreviations

For the purposes of the present document, the abbreviations given in 3GPP TR 21.905 [1], in 3GPP TS 29.163 [4], in
ITU-T E.164 [9], in ITU-T Q.1912.5 [20], and the following apply. An abbreviation defined in the present document
takes precedence over the definition of the same abbreviation, if any, in any other of the specification listed above.

IW-MGW   Interworking Media Gateway Function

4   Network characteristics

4.1   Key characteristics of ISUP/BICC based CS Domain

The 3GPP CS domain uses either BICC Capability Set 2 (see ITU-T Q.1902.1 to Q.1902.6 [19] and Q.765.5 [17]) with
3GPP specific extensions, as specified for the 3GPP Nc interface in 3GPP TS 29.205 [5], or ISUP (see ITU-T Q.761 to
Q.764 [15]), as signalling protocol.

If BICC is used as signalling protocol, the 3GPP Nb interface, as specified in 3GPP TS 29.414 [7] and 3GPP TS 29.415
[8], is used for the user plane transport. If ISUP is applied as signalling protocol, TDM transport of the user plane is
applied.

4.2   Key characteristics of external SIP-I network

The external SIP-I network applies SIP, IETF RFC 3261 [32], with ISUP encapsulated according to IETF RFC 3204
[31], as specified in ITU-T Q.1912.5 [20] Profile C. The SIP Signalling Profile defined for Profile C in Annex C of
Q.1912.5 is applied.

The network uses either IPv4 (IETF RFC 791 [24]) or IPv6 (IETF RFC 2460 [27]).

For the transport of most media, RTP (IETF RFC 3550 [36]) over UDP (IETF RFC 768 [25]) is applied.

For the transport of control signalling, UDP (IETF RFC 768 [25]), TCP (IETF RFC 793 [26]) or SCTP (IETF RFC 2960 [29]) can be used.

5 Interworking reference model

Figure 5.1 details the interworking reference model for the present specification
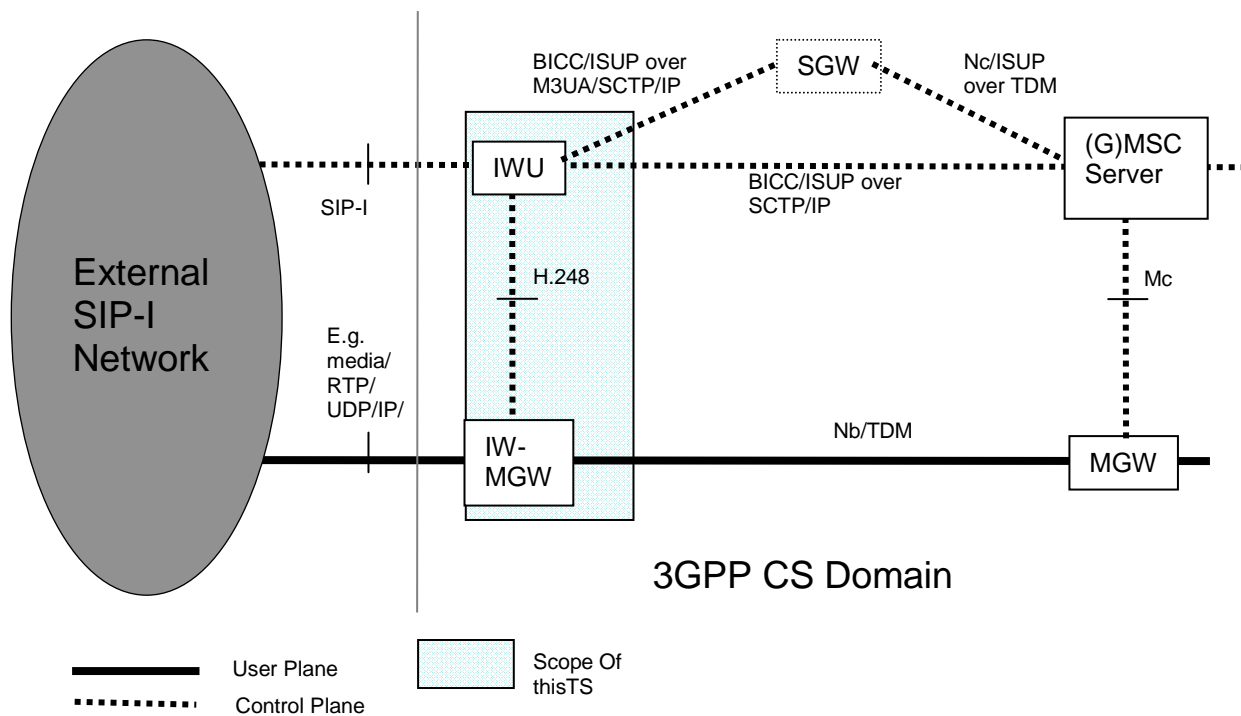
NOTE 1: If needed a SGW is applied for conversion (both ways) of the transport level between the IWU and the (G)MSC. The SGW may be implemented as a stand-alone entity or it may be located in another entity of the CS Domain. A SGW function is not required for certain signalling transports, where M3UA+SCTP+IP is used in CS network and IWU. The implementation options are not further discussed in the present document.

NOTE 2: The IWU is a logical function that may reside with other 3GPP logical functions in the same physical nodes, e.g. in an (G)MSC Server. The figure shows only the logical separation.

Figure 5.1: interworking reference model

6 Control plane interworking

6.1 General

The following sub-clauses define the signalling interworking between the Bearer Independent Call Control (BICC) or ISDN User Part (ISUP) protocols and Session Initiation Protocol (SIP) with its associated Session Description Protocol (SDP) and encapsulated ISUP at an IWU.

The IWU shall act as a Type A or Type B exchange (ITU-T Q.764 [15]) for the purposes of ISUP and BICC compatibility procedures.

The BICC/ISUP capabilities or signalling information defined for national use are outside the scope of the present document.

The interworking procedures are based on ITU-T Recommendation Q.1912.5 [20] profile C. Clarifications are made within this specification on the application of Q.1912.5 profile C.

NOTE: An IWU may apply additional procedures to support interworking for national-specific capabilities.

The services that can be supported through the use of the signalling interworking are limited to the services that are supported both within the BICC or ISUP based 3GPP CS domain and the external SIP-I network. The IWU will originate and/or terminate services or capabilities that do not interwork seamlessly across domains according to the relevant protocol recommendation or specification.

Table 6.1.1 lists the services seamlessly interworked within the scope of the present document.

Table 6.1.1: Service Interworking Capabilities

Service
Speech/3.1 kHz audio
Data Calls (optional)
En bloc address signalling
Out of band transport of DTMF tones and information. (BICC only)
Inband transport of DTMF tones and information. (BICC and ISUP)
Multiple Subscriber Number (MSN)
Calling Line Identification Presentation (CLIP)
Calling Line Identification Restriction (CLIR)
Connected line presentation (COLP)
Connected line restriction (COLR)
Call Hold
Call Forwarding
Explicit Call Transfer (ECT)
User-to-User Signalling (UUS)
Call Deflection (CD)
Closed User Group (CUG)
Completion of Calls to Busy Subscriber (CCBS)
Multi-Level Precedence and Pre-emption (MLPP)
Call Waiting

The Clause 5.3.2 of ITU-T Q.1912.5 [20] describes additional general principles specific to SIP-I.

6.2 Interworking between CS Domain using ISUP signalling and external network using SIP-I signalling

6.2.1 Control Plane Interworking

The control plane between CS networks supporting ISUP and the external network using SIP-I, where the underlying network is TDM or ATM and IP respectively, is as shown in figure 6.2.1.
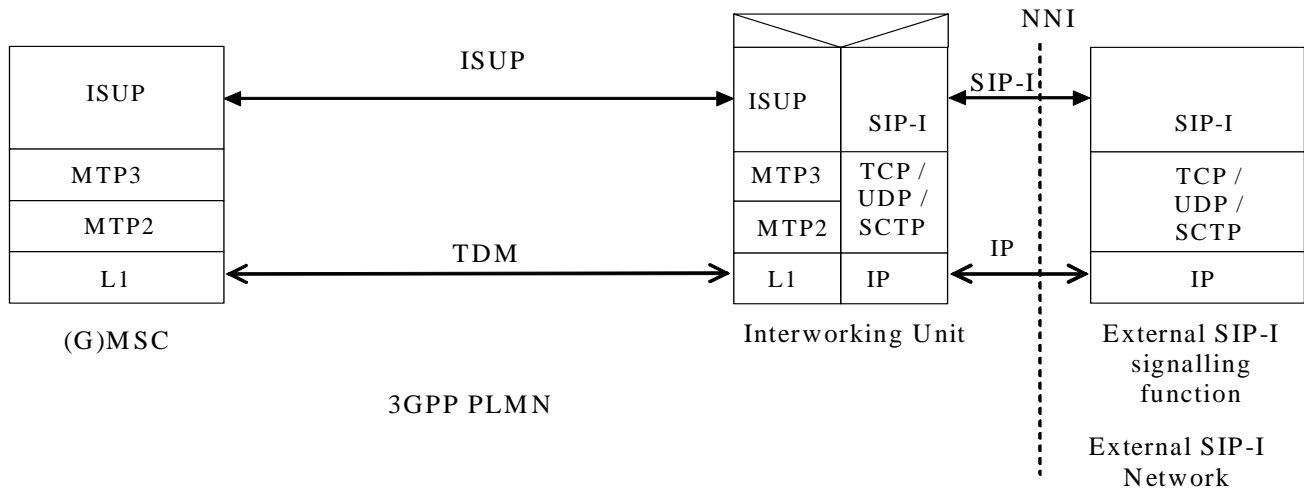
Figure 6.2.1: Control plane interworking between CS networks supporting ISUP and an external network using SIP-I signalling

6.2.2 Services performed by network entities in the control plane

6.2.2.1 Services of the IWU

The IWU shall provide the interaction, through the use of its interworking function, between the SS7 MTP3 [13] -User part information and SIP-I. The IWU shall also provide the encapsulation from the SS7 MTP3-User part information to the SIP-I ISUP MIME and de-encapsulation from this ISUP MIME to the SS7 MTP3-User part information.

6.2.2.1 Services of the external SIP-I signalling function

The external SIP-I signalling function is a remote SIP User Agent capable of processing ISUP.

6.2.3 Signalling interactions between network entities in the control plane

6.2.3.1 Signalling between the IWU and SIP-I signalling function

Signalling between the SIP-I signalling function and the IWU uses the services of IPv4 (IETF RFC 791 [24]) or IPv6 (IETF RFC 2460 [27]), transport protocol such as TCP (IETF RFC 793 [26]) or UDP (IETF RFC 768 [25]) or SCTP (IETF RFC 2960 [29]), and SIP-I.

6.2.4 SIP-ISUP protocol interworking

6.2.4.1 Incoming call interworking from SIP-I to ISUP at I-IWU

6.2.4.1.1 General

The procedures for Profile C in Clause 6 of ITU-T Q-1912-5 [20] shall be applied with the modifications provided in this Clause.

6.2.4.1.1 Interworking of received ISUP messages to SIP messages

The I-IWU receiving backwards ISUP information shall apply any interworking procedures detailed in Clause 6 of ITU-T Q-1912-5 [20] affecting parameters within the ISUP, and then proceed to encapsulate any ISUP information received (with the exception of the excluded messages detailed in 5.4.3 of ITU-T Q-1912-5 [20]) in a SIP message in a MIME body according to IETF RFC 3204 [31]. The selected SIP Header fields relating to the handling of the ISUP body shall be set as specified in Clause 5.4.1.2 of ITU-T Q-1912-5 [20].

6.2.4.1.2 Receipt of encapsulated ISUP information within SIP

On receipt of a SIP message containing encapsulated ISUP, the IWU shall de-encapsulate the ISUP message from the SIP message body. The received SIP message shall be mapped to an ISUP message and merged with the de-encapsulated ISUP message according to the rules for Profile C in Clause 6 of ITU-T Q.1912.5 [20].

NOTE 1: These precedence rules have been derived from the following principles, which can also be applied for any ISUP information not covered by the present specification:

1   Where a SIP header mapping to ISUP field(s) is possible (for example the mapping of Request-URI to Called Party Number), the SIP header is given precedence over the encapsulated ISUP value in the alignment process. (Conflicts can be caused by a possible service invocation within the external SIP-I network.)

2   De-encapsulated ISUP information overrides ISUP information derived from default values (rather than SIP information).

3   Local ISUP procedures may modify information derived from SIP or default values.

This note has been derived from text in ITU-T Q.1912.5 [20], Clause 5.4.2.

NOTE 2: There is a certain change against Q.1912.5, where the above note is formulated as normative procedures. However, this is very high level and not required if you look at the real interworking procedures later on. Therefore formulating this as a note makes much more sense.

6.2.4.1.3 Special Procedures for the Reception of SIP INVITE request

6.2.4.1.3.1 Propagation of overlap signalling toward the 3GPP CS domain

The procedures in Clause 6 of ITU-T Q.1912.5 [20] related to the propagation of overlap signalling across the I-IWU shall not be applied.

NOTE: A G-MSC acting as IWU will collect all digits required to identify the callee and not propagate the overlap signalling further. Therefore, it will not apply procedures in ITU-T Q.1912.5 [20] Clause 6 related to the propagation of overlap signalling.

6.2.4.1.3.2 Derivation of TMR, USI and HLC parameters within send IAM message

The I-IWU may choose to transcode media and shall then set the parameters according to the coding applied within the CS Domain. Otherwise, the I-IWU shall select a codec for the SIP side termination using SDP offer-answer procedures, IETF RFC 3264 [34], and shall map the SDP information of this codec to the TMR/USI/HLC parameters according to table 2a of 3GPP TS 29.163 [4]. If the information derived from this mapping matches the information in the TMR/USI/HLC parameters in the encapsulated ISUP, the TMR/USI/HLC parameters from the encapsulated ISUP should be used as they may contain additional information. If the information derived from this mapping contradicts the information in the TMR/USI/HLC parameters in the encapsulated ISUP, the TMR/USI/HLC parameters derived by the mapping shall be used.

NOTE: The procedures in this note are an amendment compared to ITU-T Q.1912.5, which simply states the TMR parameters in encapsulated ISUP shall take precedence, However this is inappropriate if an incompatible codec is selected.

6.2.4.1.3.3 Receipt of SIP INVITE without SDP offer

An IWU may reject receipt of SIP INVITE without SDP offer, otherwise the procedures in this paragraph shall be followed.

Upon receipt of the first INVITE with sufficient digits for an IAM to be sent, but without an SDP offer (see IETF RFC3264 [34]), the I-IWU shall construct an SDP offer with contents according to local policy, e.g. SDP for a G711 speech call. The IWU may use the TMR and USI parameters of the encapsulated IAM to determine the desired service and construct the SDP offer accordingly.

1) If reliable provisional responses (see IETF RFC 3262 [33]) are supported in the external SIP-I network, the I-IWU should immediately send the SDP offer within a 183 Session Progress message. The I-IWU should send the IAM upon receipt of the SDP answer and may take into account the media selected in the answer when constructing the IAM.

2) If reliable provisional responses are not supported, the I-IWU should immediately send out the IAM. The I-IWU will also include the SDP offer in the 200 OK(INVITE) SIP messages according to SIP procedures.

NOTE: The text in the preceding paragraph has been derived from ITU-T Q.1912.5 [20], Clause 6.1.1. Compared to ITU-T Q.1912.5, the recommendation has been added that for an incoming INVITE without media the TMR/USI is considered to select media. Furthermore, bullet 1 recommends to wait for the SDP answer to construct TMR, USI and BICC codec list accordingly.

6.2.4.2 Outgoing Call Interworking from ISUP to SIP-I at O-IWU

6.2.4.2.1 General

The procedures for Profile C in Clause 7 of ITU-T Q-1912-5 [20] shall be applied with the modifications provided in this Clause.

6.2.4.2.2 Sending of ISUP information to adjacent SIP nodes

The I-IWU receiving ISUP information shall apply any interworking procedures detailed in Clause 7 of ITU-T Q-1912-5 [20] affecting parameters within the ISUP, and then proceed to encapsulate any ISUP information received (with the exception of the excluded messages detailed in 5.4.3 of ITU-T Q-1912-5 [20]) in a SIP message in a MIME body according to IETF RFC 3204 [31]. SIP Header fields relating to the handling of the ISUP body shall be set as specified in 5.4.1.2 of ITU-T Q-1912-5 [20].

NOTE: The text in the preceding paragraph has been derived from ITU-T Q.1912.5 [20], Clause 5.4.1.

For a basic call setup, the SIP message used to encapsulate the ISUP message shall be the SIP message that was first triggered to be sent from the IWU as a result of the interworking specified in Clause 7 of ITU-T Q-1912-5 [20]]. As an example, this means that an ISUP IAM will be encapsulated within the INVITE message that is sent out from the O-IWU. For the ISUP messages listed in Table 1 of ITU-T Q.1912.5 [20], the special procedures in Clause 5.4.3 of ITU-T Q.1912.5 are applicable.

NOTE: The text in the preceding paragraph has been derived from ITU-T Q.1912.5 [20], Clause 5.4.1.3.

6.2.4.2.3 Receipt of encapsulated ISUP information within SIP-I

On receipt of a SIP message containing encapsulated ISUP, the IWU shall de-encapsulate the ISUP message from the SIP message body. The received SIP message shall be mapped to an ISUP message and merged with the de-encapsulated ISUP information according to the rules in in Clause 7 of ITU-T Q-1912-5 [20].

NOTE: The text in the preceding paragraph has been derived from ITU-T Q.1912.5 [20], Clause 5.4.2.

NOTE: These precedence rules have been derived from the following principles, which can also be applied for any ISUP information not covered by the present specification:

1   Where a SIP header mapping to ISUP field(s) is possible (for example the mapping of Request-URI to Called Party Number), the SIP header is given precedence over the encapsulated ISUP value in the alignment process. (Conflicts can be caused by a possible service invocation within the SIP network.)

2   De-encapsulated ISUP information overrides ISUP information derived from default values (rather than SIP information).

3   Local ISUP procedures may modify information derived from SIP or default values.

This Note has been derived from text in ITU-T Q.1912.5 [20], Clause 5.4.2

NOTE   There is a certain change against Q.1912.5, where the above note is formulated as normative procedures. However, this is very high level and not required if you look at the real interworking procedures later on. Therefore formulating this as a Note makes much more sense.

6.2.4.2.4 Special procedures related to outgoing INVITE

6.2.4.2.4.1 Overlap Signalling

The O-IWU does not need to support procedures related to overlap signalling in Clause 7 of ITU-T Q-1912-5 [20].

NOTE: No overlap signalling is used in a 3GPP CS domain.

6.2.4.2.4.2 Coding of encapsulated ISUP IAM parameters in outgoing INVITE

The O-IWU may choose to transcode media, or attempt to interwork media without transcoding. If the O-IWU transcodes, it should set the TMR/USI/HLC parameters according to the codec applied in the SIP-I network. Otherwise, it should provide the TMR/USI/HLC parameters as received in the incoming IAM. If the I-IWU offers several codecs within SDP, it should set the TMR/USI/HLC parameters according to the preferred codec, applying the rules above for this codec.

NOTE: ITU-T Q.1912.5 [20] does not describe the relationship between TMR/USI/HLC and SDP codec negotiation.

6.2.4.2.4.3 Media offered in SDP of outgoing INVITE

The O-IWU should offer codecs known to be supported within the external SIP-I network.

6.3 Interworking between CS Domain using BICC signalling and external network using SIP-I signalling

6.3.1 Control Plane Interworking

The control plane between CS networks supporting BICC and the external network using SIP-I, where the underlying network is SS7 and IP respectively, is shown in figures 6.3.1 and 6.3.2 in accordance to TS 29.202 [37].
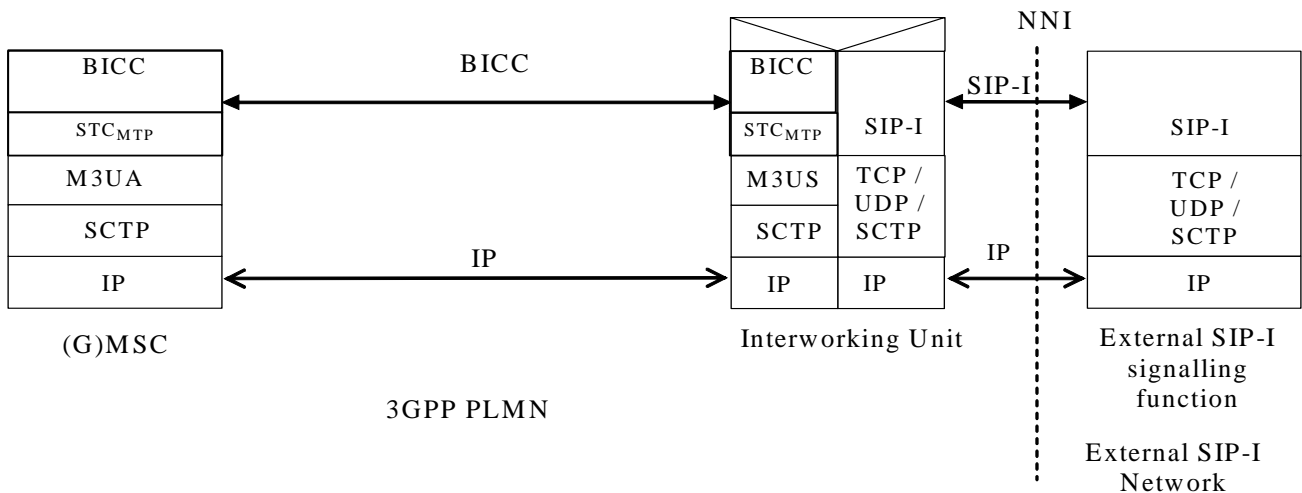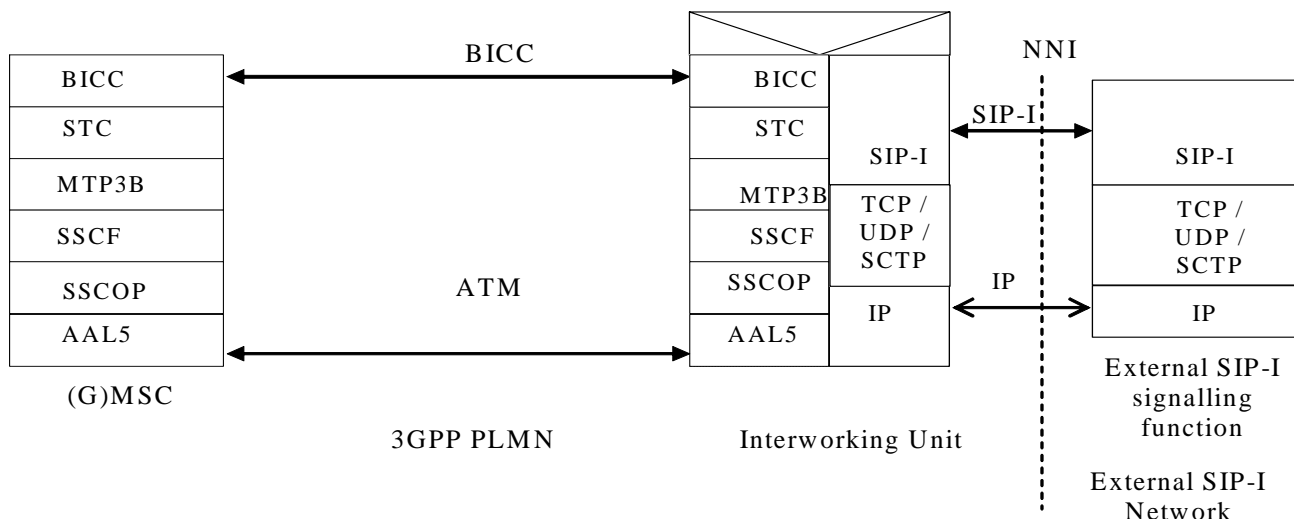

Figure 6.3.1: Control plane interworking between CS networks supporting BICC/IP and an external network using SIP-I signalling

Figure 6.3.2: Control plane interworking between CS networks supporting BICC/ATM and an external network using SIP-I signalling

For signalling interactions between network entities in the control plane see Clause 6.2.3.

6.3.2   SIP-BICC protocol interworking

6.3.2.1   General

Specific rules for handling of the APM mechanism have been added, which are not specified in ITUT Q.1912.5 [20].

The procedures in Clause 6.2.4 shall be applied with the modifications provided the present Clause.

The text in Clause 6.2.4 is to be understood as follows:

-     Where "ISUP" is mentioned, this shall be understood as BICC. As an exception, references to ISUP encapsulated within SIP-I shall be understood without modification, i.e. they still refer to ISUP rather than BICC.

6.3.2.2   Incoming call interworking from SIP to BICC at I-IWU

An APM messages received from the CS side (see ITU-T Q.765 [16]) that relates to the BICC APM user (see ITU-T Q.765.5 [17]) shall not be encapsulated in any triggered SIP message.

6.3.2.3   Outgoing Call Interworking from BICC to SIP at O-IWU

If an IAM message is received, the APM information elements (see ITU-T Q.765 [16]) relating to the BICC APM user (see ITU-T Q.765.5 [17]) shall be removed before the IAM message is encapsulated in the triggered SIP INVITE message.

An APM messages received from the CS side (see ITU-T Q.765 [16]) that relates to the BICC APM user (see ITU-T Q.765.5 [17]) shall not be encapsulated in any triggered SIP message.

6.4   Supplementary services

6.4.1   Special procedures for supplementary service interworking

The supplementary services described in Table 6.1.1 are interworked by using the parameters of the (de)encapsulated ISUP. No other interworking is required, except if otherwise described within the clauses below.

6.4.2   Interworking of CLIP/CLIR supplementary service

At the O-IWU: the service shall be supported by encapsulation.

At the I-IWU: ITU-T Q.1912.5 [20], Annex B.1, shall apply.

6.4.3   Interworking of Call Hold (HOLD) supplementary service

The Profile C procedures in ITU-T Q.1912.5 [20], Clause B10, shall be followed.

6.4.4   Interworking of Completion of Calls to Busy Subscriber (CCBS) supplementary service to SIP networks

The Profile C procedures of ITU-T Q.1912.5 [20], Clause B.11 shall be applied.
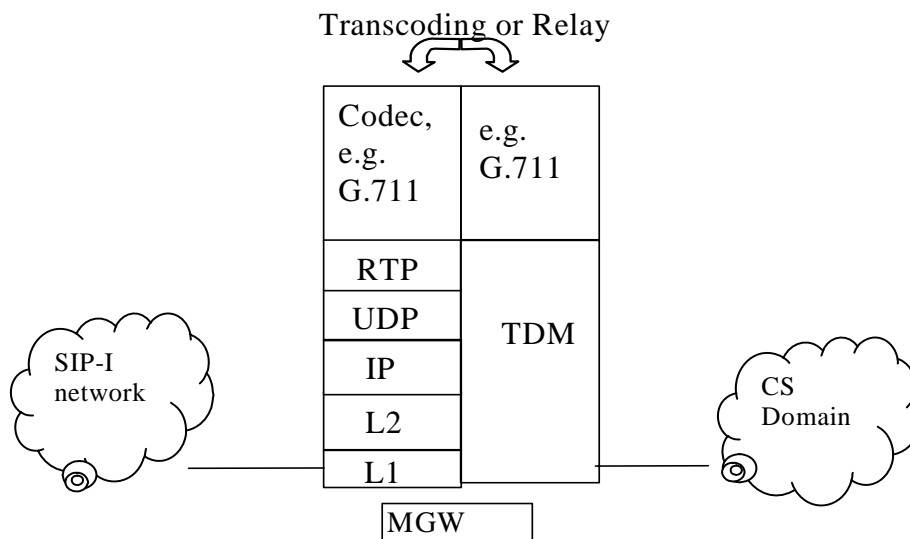
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7   User plane Interworking

7.1   ISUP based CS Domain

Figure 7.1.1 shows the user plane protocol stacks within the SIP-I network and an ISUP based CS Domain.

Apart from speech codecs, data call related codecs, e.g. as listed in Table 2a of 3GPP TS 29.163 [4], may be used.

Figure 7.1.1: User Plane Interworking and ISUP based CS Domain

7.2   BICC based CS Domain

Figure 7.2.1 shows the user plane protocol stacks within the SIP-I network and an BICC based CS Domain.

Apart from speech codecs, data call related codecs, e.g. as listed in Table 2a of 3GPP TS 29.163 [4], may be used.

Within the CS domain, the Nb interface as specified in 3GPP TS 29.414 [7] and 3GPP TS 29.415 [8] is used.

If the same codec is used on both sides, no transcoding is required. If no transcoding is performed, the procedures in Clause 8.1.1 of 3GPP TS 29.163 [4] shall be applied to interwork the Nb framing Protocol, as specified in 3GPP TS 29.415 [8], with RTP, as specified in IETF RFC 3550 [36].

Transcoding or Relay
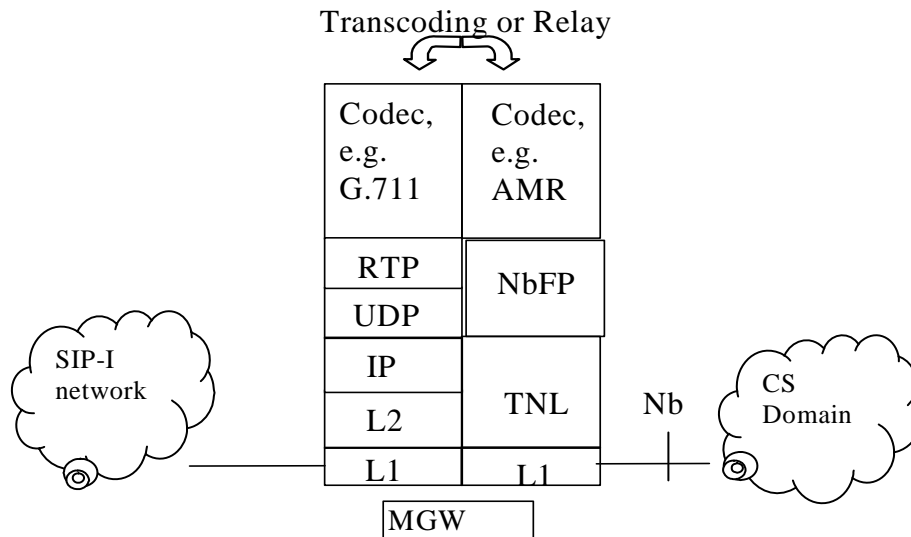
Figure 7.2.1: User Plane Interworking for BICC based CS Domain

8 Interactions between control node and MGW

8.1 Overview

ITU-T H.248.1 [11] with 3GPP specific extensions as detailed in 3GPP TS 29.332 [6] shall be applied as signalling protocol between IWU and MGW.

The present specification describes the signalling procedures between IWU and MGW and their interaction with BICC/ISUP and SIP signalling in the control plane, and with user plane procedures. 3GPP TS 29.332 [6] maps these signalling procedures to H.248 messages and defines the required packages and parameters.

8.2 Signalling interactions

The procedures in Clause 9.2 of TS 29.163 shall be applied with modifications provided in the present Clause.

8.2.1 Backward through-connection for outgoing call interworking from BICC to SIP at O-IWU.

For the outgoing call interworking from BICC to SIP at O-IWU, the IWU shall not request the MGW to provide an awaiting answer indication (ringing tone) to the calling party, as specified in Clauses 9.2.3.1.5, 9.2.3.2.5 and 9.2.3.3.5 of 3GPP TS 29.163 [4]. The O-IWU shall instead request the MGW to backward-through connect using the "Change-IMS Through-Connection" procedure when the trigger conditions in those clauses apply.

8.2.2 Handling of RTP telephone events

The procedures in Clause 9.2.8 of 3GPP TS 29.163 [4], are applicable with the following modifications:

It depends on the characteristics of the external SIP-I network if DTMF is transported within the codec or according to IETF RFC 4733 [28]. SIP-I transports DTMF within the RTP telephony event according to RFC 4733 [28] if the RTP telephony event payload type is negotiated between the peer SIP-I entities; alternatively, if the RTP telephony event payload has not been negotiated, SIP-I may transport the DTMF in-band ("within the codec") when the selected codec allows transparent DTMF transport.

The control node indicates the transport mode of DTMF to the MGW as follows in the "Configure IMS Resources" procedure or "Reserve IMS Connection Point and Configure Remote Resources" procedure (signal 1 in figure 48 or figure 49 of 3GPP TS 29.163 [4]):

-    If the control node does not configure the RTP telephony event payload at the MGW, then the Detect DTMF procedure means in-band ("within the codec") detection and sending of DTMF.

-    If the control node configures the RTP telephony event payload at the MGW, then the Detect DTMF procedure means extracting and sending DTMF within the RTP payload type according to IETF RFC 4733 [28].

Editor's Note: The above reference contains sequences that require the MGW to forward partial digits to the Server. It is not yet agreed if this shall be permitted or if the MGW shall only forward complete and valid digits.

8.3   Signalling procedures

The procedures in Clause 9.3 of 3GPP TS 29.163 shall be applied with modifications provided in the present Clause.

The support of the "Send TDM Tone", "Stop TDM Tone", "Send Tone" and "Stop Tone" procedures is optional.

Annex A (normative):
Codec Negotiation between a BICC based CS Domain and an external SIP-I network

The procedures in Annex B of TS 29.163 [4] are applicable when the IWU receives from the external SIP-I based network an SDP offer without the 3GPP_OoBTC_Indicator as defined in 3GPP TS 29.231 [38].

If the IWU receives from the external SIP-I based network a SIP request with an SDP offer containing a codec list with the 3GPP_OoBTC_Indicator, the IWU may apply the procedures in accordance to subclause 6.6.2 of 3GPP TS 29.235 [39].

When the IWU sends a SIP request with an SDP offer towards the external SIP-I based network, the IWU may include the 3GPP_OoBTC_Indicator in accordance to subclause 6.6.3 of 3GPP TS 29.235 [39].

Annex B (informative):
Change history

Change history								
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**		**Old**	**New**
01/09/2006	CT3#41	C3-060509			Agreed Skeleton of TR			0.0.0
15/10/2006					Proposed update of the TR on CT3 email reflector NOT discussed or agreed by CT3, but put onto the 3GPP web server drafts folder by mistake.		0.0.0	0.1.0
08/11/2006	CT3#42	C3-060877			Includes the following TDOCs agreed by CT3: C3-060761, C3-060667, C3-060837, C3-060763, C3-060765, C3-060860, C3-040840, C3-040841, C3-040769, C3-060861, C3-060847, C3-060813, C3-060678, C3-060679		0.0.0	0.2.0
22/02/2007	CT3#43	C3-070274			Includes the following TDOCs agreed by CT3: C3-070105, C3-070106, C3-070249, C3-070258		0.2.0	0.3.0
28/02/2007	TSG#35	CP-070100			Editorial update by MCC for presentation to TSG CT for information		0.3.0	1.0.0
19/04/2007	CT3#43bis				Includes the following TDOCs agreed by CT3: C3-070355, C3-070373, C3-070374		1.0.0	1.1.0
15/05/2007	CT3#44				Includes the following TDOC agreed by CT3: C3-070523		1.1.0	1.2.0
23/05/2007	TSG#36	CP-070427			Editorial update by MCC for presentation to TSG CT for approval		1.2.0	2.0.0
06/2007	TSG#36	CP-070427			Approved to be made v7.0.0 and put under change control		2.0.0	7.0.0
12/2008	TSG#42				Upgraded to v8.0.0 due to simple upgrade without no technical change		7.0.0	8.0.0
05/2009	TSG#44	CP-090350	001	2	Codec negotiation when interworking with BICC based CS CN		8.0.0	8.1.0

History

Document history		
V8.0.0	February 2009	Publication
V8.1.0	June 2009	Publication