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Foreword

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1 Scope

The present document provides a study into providing a SIP-I based Nc interface (G)MSC-S - (G)MSC-S as an alternative to the existing BICC definition. The document covers the functional description, network architecture and protocol definition for the SIP-I based solution. The contents of this report when stable shall be moved into a Technical Specification 3GPP TS ab.cde along with modifications to existing specifications.

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 29.205: "Application of Q.1900 series to Bearer Independent circuit-switched network architecture; Stage 3".

[4] ITU-T Recommendation Q.1902.1: "Bearer Independent Call Control CS2 Functional Description".

[5] ITU-T Recommendation Q.1902.2: "Bearer Independent Call Control CS2 General Functions of Messages and Signals".

[6] ITU-T Recommendation Q.1902.3: "Bearer Independent Call Control CS2 Formats and Codes".

[7] ITU-T Recommendation Q.1902.4: "Bearer Independent Call Control CS2 Basic Call Procedures".

[8] ITU-T Recommendation Q.1902.5: "Exceptions to the Application Transport Mechanism in the Context of Bearer Independent Call Control".

[9] ITU-T Recommendation Q.1902.6: "Generic Signalling Procedures and Support of the ISDN User Part Supplementary Services with the Bearer Independent Call Control Protocol".

[10] 3GPP TS 29.232: "Media Gateway Controller (MGC) - Media Gateway (MGW) interface; Stage 3".

[11] ITU-T Recommendation H.248.1 (05/2002): "Gateway control protocol". Version 2.

[12] 3GPP TS 29.414: "Core Network Nb data transport and transport signalling".

[13] 3GPP TS 29.415: "Core network Nb data transport and transport signalling".

[14] 3GPP TS 25.414: "UTRAN Iu interface data transport and transport signalling".

[15] 3GPP TS 25.415: "UTRAN Iu interface user plane protocols".

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

[17] 3GPP TS 23.153: "Out of band transcoder control; Stage 2".

[18] IETF RFC 3267: "Real-Time Transport Protocol (RTP) Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs".

[19] ITU-T Recommendation Q.1970: " BICC IP Bearer Control protocol".

[20] 3GPP TS 29.332: "Media Gateway Control Function (MGCF) - IM Media Gateway (IM-MGW); Mn interface".

[21] 3GPP TS 23.228: "IP Multimedia Subsystem (IMS); Stage 2".

[22] 3GPP TS 23.172: "Technical realization of Circuit Switched (CS) multimedia service; UDI/RDI fallback and service modification; Stage 2".

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

[24] 3GPP TS 24.229: "Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3".

[25] 3GPP TS 29.163: "Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks".

[26] IETF RFC 2327: "SDP: Session Description Protocol".

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

[28] IETF RFC 3551: "RTP Profile for Audio and Video Conferences with Minimal Control".

[29] IETF RFC 3555: "MIME Type Registration of RTP Payload Formats".

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

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

[32] 3GPP TS 33.210: "Network Domain Security; IP network layer security".

[33] 3GPP TS 26.235: "Packet switched conversational multimedia applications; Default codecs".

[34] IETF RFC 4040: "RTP Payload Format for a 64 kbit/s Transparent Call".

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

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

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

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

[39] 3GPP TS 23.108: "Mobile radio interface layer 3 specification core network protocols; Stage 2".

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

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

[42] IETF draft-ietf-avt-rtp-amr-bis-06 (September 2006): "Real-Time Transport Protocol (RTP) Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs".

[43] 3GPP TS 26.103: "Speech codec list for GSM and UMTS".

[44] 3GPP TS 23.091: "Explicit Call Transfer (ECT) Supplementary Service; Stage 2".

[45] 3GPP TR 29.863: "Feasibility Study for the multimedia interworking between the IM CN subsystem and CS Networks".

[46] 3GPP TS 29.164: "Interworking between the 3GPP CS domain with BICC or ISUP as signalling protocol and external SIP-I networks".

[47] 3GPP TS 23.202: "Circuit switched data bearer services".

[48] 3GPP TS 29.007: "General requirements on Interworking between the Public Land Mobile Network (PLMN) and the Integrated Services Digital Network (ISDN) or Public Switched Telephone Network (PSTN)".

[49] 3GPP TS 23.146: "Technical realization of facsimile group 3 non-transparent".

[50] ITU-T Recommendation V.152: "Procedures for supporting Voice-Band Data over IP Networks".

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

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

[53] IETF RFC 2976 (October 2000): "The SIP INFO method".

[54] IETF RFC 3204 (December 2001): "MIME media types for ISUP and QSIG Objects".

[55] IETF RFC 3323 (November 2002): "A Privacy Mechanism for the Session Initiation Protocol (SIP)".

[56] IETF RFC 3325 (November 2002): "Private Extensions to the Session Initiation Protocol (SIP) for Network Asserted Identity within Trusted Networks".

[57] IETF RFC 3326 (December 2002): "The Reason Header Field for the Session Initiation Protocol (SIP)".

[58] IETF RFC 4566 (July 2006): "SDP: Session Description Protocol".

[59] 3GPP TS 29.202: "SS7 Signalling Transport in Core Network; Stage 3".

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

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

[62] IETF RFC 3640: "RTP Payload Format for Transport of MPEG-4 Elementary Streams".

[63] IETF RFC 4103: "RTP Payload for Text Conversation".

[64] 3GPP TS 48.008: "Mobile Switching Centre – Base Station system (MSC – BSS) interface layer 3 Specification".

[65] 3GPP TS 23.009: "Handover procedures".

[66] 3GPP TS 24.008: "Mobile radio interface Layer 3 specification; Core network protocols; Stage 3".

[67] 3GPP TS 43.051: "GSM/EDGE Radio Access Network (GERAN) overall description; Stage 2".

[68] 3GPP TR 29.814: "Feasibility Study on Bandwidth Savings at Nb Interface with IP transport".

[69] IETF RFC 3984: "RTP Payload Format for H.264 Video".

[70] 3GPP TS 23.002: "Network Architecture".

[71] NICC ND1017:2006/07: "TSG/SPEC/017; Interworking between Session Initiation Protocol (SIP) and UK ISDN User Part (UK ISUP)".

[72] IETF RFC 4694: "Number Portability Parameters for the "tel" URI".

[73] 3GPP TS 23.066: "Support of Mobile Number Portability (MNP); Technical Realization; Stage 2".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in TR 21.905 [1] and the following apply. A term defined in the present document takes precedence over the definition of the same term, if any, in TR 21.905 [1].

Intermediate node: a 3GPP node that may be between either two other 3GPP nodes (e.g., MSC), or between a 3GPP node (e.g. GMSC) and the external network, or between two external networks (e.g., GMSC, MSC/IWF, in the case of incoming mobile call being immediately forwarded back out to an external network).

3.2 Symbols

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

Iu	Interface between the RNS and the core network. It is also considered as a reference point.
Mc	Interface between the server and the media gateway.
Nb	Interface between media gateways.
Nc	The NNI call control interface between (G)MSC servers.

3.3 Abbreviations

For the purposes of the present document, the abbreviations given in TR 21.905 [1] and the following apply. An abbreviation defined in the present document takes precedence over the definition of the same abbreviation, if any, in TR 21.905 [1].

BCF	Bearer Control Function
BICC	Bearer Independent Call Control
BICN	Bearer Independent Core Network
CS	Circuit Switched
DSP	Digital Signal Processing
GERAN	GSM/EDGE Radio Access Network
(G)MSC-S	(Gateway) MSC Server
IAM	Initial Address Message
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPBCP	IP Bearer Control Protocol
IWF	Interworking Function
MGCF	Media Gateway Control Function
MGW	Media Gateway
MSC-S	MSC Server
MSC/IWF	IWF at call control layer toward external network (see 3GPP TS 29.007)
NNI	Network-Network interface
OoBTC	Out of Band Transcoder Control
PRACK	Provisional Response Acknowledgement
RTO	Remote Transcoder Operation
RTP	Real-Time Transport Protocol
SCTP	Stream Control Transmission Protocol
SCUDIF	Service Change and UDI Fallback
SIP	Session Initiation Protocol
SDP	Session Description Protocol
TCP	Transmission Control Protocol
TDM	Time-Division Multiplexing
TFO	Tandem Free Operation
TrFO	Transcoder Free Operation
UDP	User Datagram Protocol
UTRAN	UMTS Terrestrial Radio Access Network

4 Overview

4.1 Background

In 3GPP R4, the coupling between the Bearer and Control plane in the Circuit Switched (CS) core network was broken with the introduction of the Bearer Independent Core Network (BICN). The BICN allowed for the definition of a separate control plane that could control the bearer plane independently of the transport technology used for the bearer, thus opening the possibility to use ATM or IP transport for CS traffic, as well as TDM.

The specification of this architecture and the associated required functionality and call flows can be found in 3GPP TS 23.205 [2]. Within this architecture, three interfaces were identified;-

- The Nc interface, this being the interface between (G)MSC-S and (G)MSC-S. This interface is further defined in 3GPP TS 29.205 [3] and is based on BICC as defined in ITU-T recommendations Q.1902.1 to Q.1902.6 (see [4] to [9]).
- The Mc interface, this being the interface between (G)MSC-S and MGW. This interface is further defined in 3GPP TS 29.232 [10] and is based on the ITU-T recommendation H.248.1 [11].
- The Nb interface, this being the interface between MGW and MGW. This interface is further defined in 3GPP TS 29.414 [12] and 3GPP TS 29.415 [13] and is based on the transport layer of the Iu interface defined in 3GPP TS 25.414 [14] and 3GPP TS 25.415 [15].

Within the context of this TR, ,an intermediate MSC is any MSC that interworks a call from an originating side to a terminating side, e.g., gateway MSC, visited MSC during call forwarding or an anchor MSC during handover.

The architecture is shown below.

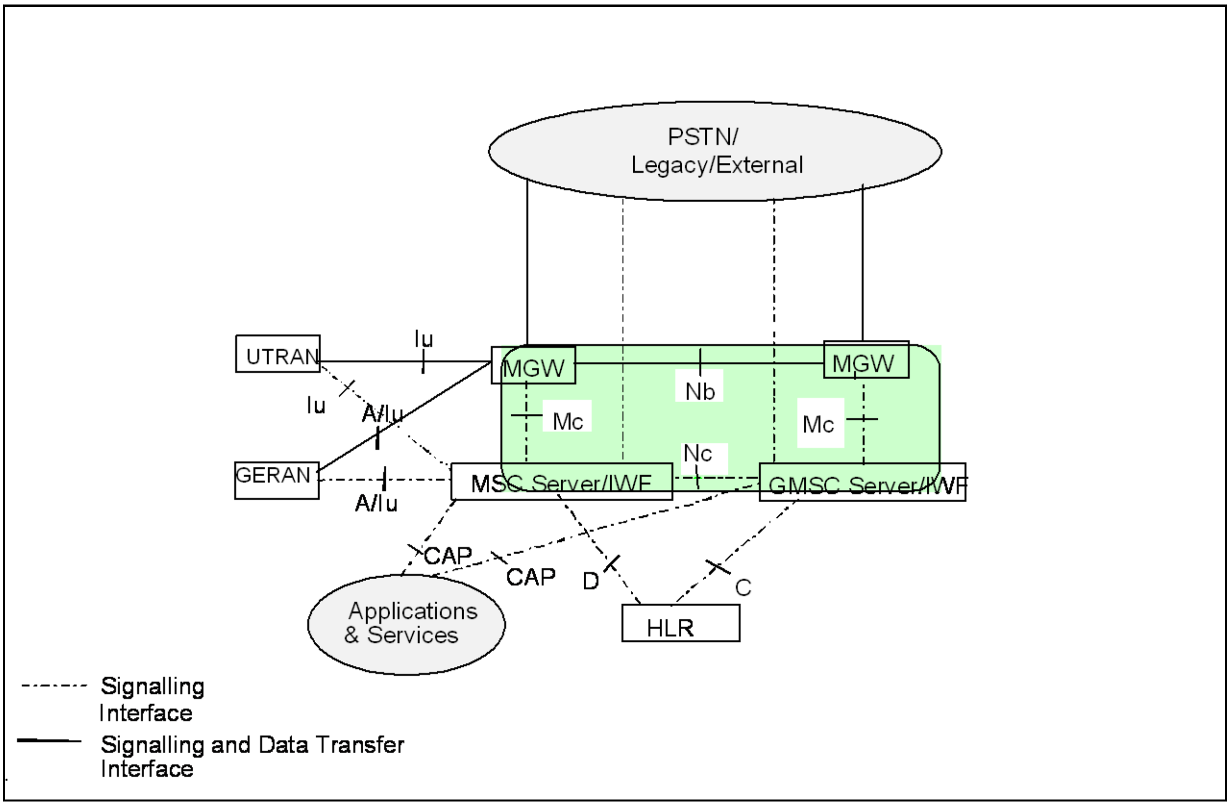

Figure 4.1.1: CS core network logical architecture

In section 5.1.2.2 of 3GPP TS 23.205 [2] on the Nc interface, the statement that 'Any suitable call control protocol may be used over the Nc interface (e.g. BICC)' can be found. 3GPP TS 29.205 [3] currently only defines the use of BICC on the Nc interface. However, other possibilities exist.

Within this TR, the possibility to use a different protocol on the interface between (G)MSC-S's is considered. The protocol that would act as the basis for the work is SIP-I as defined in ITU-T recommendation Q.1912.5 [16]. The use of SIP-I would be specific to the use of IP in the Bearer plane and so, strictly, the BICN architecture could not be considered to be Bearer Independent if SIP-I was used on the Nc interface.

The use of IP in the bearer plane is a growing trend within the telecoms industry, and the associated trend in non-3GPP networks is for SIP-I to be used as the control plane protocol. As a result, the mobile industry dependence on BICC may increasingly create 'islands' of mobile networks that have to be interconnected via SIP-I controlled transit networks, or connected via those SIP-I based transits to SIP-I based fixed line operator networks. However, it is not easy for the mobile operator to move away from BICC as it has mobile industry specific extensions associated with it for the support of Out Of Band Transcoder Control (OoBTC) which is key to the success of Transcoder Free Operation (TrFO) as defined in 3GPP TS 23.153 [17].

Three possible solutions exist to resolve this misalignment;-

1. Redefine the call control protocol for BICN as SIP-I, with suitable extensions to SIP-I to support TrFO.

2. Define the interworking of BICC to SIP-I, and define TrFO extensions for SIP-I to traverse a transit network.

3. Maintain the current TDM interconnects between operators.

It should be noted that many transit networks already transport native TDM traffic over IP, and so in option 3, whilst maintaining TDM interconnects might technically seem like an option, there are in fact conversion points between TDM and IP and back again already in place. Hence moving towards an IP end-to-end interconnect will result in considerably less delay, less transcoding and a reduction in other bearer manipulation.

Option 2 would have the advantage of not requiring existing BICC networks to alter their implementation, whilst also accommodating TrFO through the extensions to the SIP-I profile. However, there is still a requirement for an Interworking Function (IWF) at the edge of the mobile network, and a considerable amount of standards effort to define that IWF to maintain the OoBTC signalling across the conversion point. Perhaps more importantly, there is a need to define how OoBTC would operate over SIP-I.

Since OoBTC functioning would need to be defined for SIP-I for Option 2 to be successful, this would make Option 1 possible as well, where Option 1 would remove the need for an IWF from BICC to an external SIP-I network. The decision of whether to go with Option 1 or Option 2 is really one for an individual operator to make based on what they have deployed. However, the net result is that an OoBTC enabled SIP-I would need to be defined, and this in turn amounts to equivalent functionality to that required on Nc.

4.2 Functional Impacts

In order for SIP-I to be used as the control plane protocol for the support of OoBTC enabled mobile to mobile calls, it is required to define the way in which codec negotiation, out of band takes place at call initiation and during the call. Also, the use of SIP-I on the Nc interface requires SIP-I to support Nc interface functionality. However, defining the protocol to be used on the Nc interface may permit a solution that ignore functions that either are not required with SIP-I that are defined in BICC or that have been found to be redundant, however care must be taken that interworking and backward compatibility are maintained. Conversely, SIP-I may offer the ability to support additional functions that are not currently defined for BICC based call control models.

One such area is likely to be in the handling of the User Plane Framing Protocol. Nc interface based on BICC assumes the default user plane utilises the Iu-UP framing on the Nb interface but, in moving to SIP-I such support for the Iu Framing does not exist within the SIP-I profile Q.1912.5 Profile C. Hence SIP-I can either be extended, or alternatively the use of Iu Framing on the Nb interface can be reviewed. This would lead to the likely adoption of RTP transport for the bearer plane (see IETF RFC 3267 [18]), which would in turn bring alignment between the Nb interface and the Mb interface.

Similarly, BICC utilizes the IP Bearer Control Protocol (ITU-T recommendation Q.1970 [19]) for transport of media characteristics when the bearer is IP. IPBCP is initiated at the MGW and transported transparently in the Mc and Nc interfaces. However, it is really a container for transport of SDP, and so when using SIP-I on Nc, the SDP can be included within the message body itself. This would make it logical to transport SDP directly within H.248 commands on the Mc interface, making the Mc interface look a lot more like the Mn interface (see 3GPP TS 29.332 [20]).

The result of the adoption of a SIP-I based call control interface may promote functional alignment between the functions of the MSC-S/MGW in the BICN architecture and the functions of the MGCF/IM-MGW in the IMS architecture (see 3GPP TS 23.228 [21]).

This TR investigates the functional impact and the required specification work for the support of SIP-I based call control of IP bearers in the Circuit Switched Core Network.

5 Basic Principles

5.1 Introduction

The introduction of a SIP-I based Nc interface has implications on a number of different aspects within the Circuit Switched Core Network. This section highlights the impacted areas and describes the basic principles for providing the same functional capabilities as provided by a BICC-based Nc Interface.

Specifically, impacted areas are:

- The implementation of a SIP-I based Nc Interface shall include the capability to provide the same functionality that a BICC based Nc interface provides. Therefore a SIP-I profile shall be specified in order to provide this functionality.

- Features that are currently optional in TS 23.205 should be optional in SIP-I and features that are currently mandated in TS 23.205 shall remain mandatory; any proposal to change to this status needs to be agreed with suitable justification.

- Minimising the interworking function is also highly desired to facilitate interoperation with existing SIP-I networks. This may be achieved by specific profiling options in the profile or inclusion of certain procedures on the Nc interface that are only required by the external SIP-I network; evaluated on a case by case basis

- The 3GPP SIP-I profile should clearly describe the minimum functionality required within a closed 3GPP network and any potential extensions required to interoperate with external SIP-I networks.

- The signalling transport for a SIP-I based Nc interface may be impacted. An appropriate single transport layer protocol is required to be specified.

- The IP Bearer Control Protocol (IPBCP) ITU-T Recommendation Q.1970 [19] is currently used on a BICC based Nc Interface in order to transport the media characteristics of an IP bearer. An appropriate mechanism to transport the media characteristics shall be specified for a SIP-I based Nc interface.

- Currently a BICC based Nc Interface specifies four alternative call establishment mechanisms in 3GPP TS 23.205 [2]. These are Fast Forward Bearer Establishment, Delayed Forward Bearer Establishment, Fast Backward Bearer Establishment, and Delayed Backward Bearer Establishment. SIP-I based Nc interface shall be defined only to support the Call Establishment Models that are relevant to network operation with IP bearers. This may be a subset of the currently defined call establishment models.

- The BICC based Nc Interface currently supports the ability to provide codec negotiation which in turn allows the OoBTC/TrFO functionality defined in 3GPP TS 23.153 [17] and SCUDIF 3GPP TS 23.172 [22] and TFO harmonisation. This functionality shall be supported on a SIP-I based Nc interface, therefore a mechanism for codec negotiation on the Nc interface shall be specified. Furthermore, the codecs that are mandatory and optional on Nc shall be specified.

- Existing services (e.g. Circuit-Switched Data, DTMF, etc) shall be specified for a SIP-I based Nc interface.

- The implementation of a SIP-I based Nc interface shall interwork with the existing BICC based Nc interface. In addition interworking with additional interfaces currently supported by the BICN shall be supported (e.g. 3GPP IMS SIP).

- The architecture shall conform to 3GPP TS 23.002 [70]. Only logical 3GPP nodes need to be described and therefore Call Mediation Nodes, SIP Proxies or Transit Servers are not part of the 3GPP CS PLMN and shall not be included in the study of SIP-I on Nc. Scenarios described in 3GPP TS 23.205 where a node can remove its

MGW due to call forwarding optimisations (Bearer Redirection) are valid but not considered under to be within the definition of CMN.

5.2 Definition of a 3GPP SIP-I Profile

5.2.1 General

General principles of the 3GPP SIP-I profile require further discussion in terms of requirements of the Nc interface and requirements to provide full interworking to external SIP-I networks. Both requirements are equally important in the study however the study should evaluate the option of separate interworking requirements (additional to Nc requirements) being interworked at the intermediate node or all Q1912.5 principles applicable to external SIP-I networks being fully applied within 3GPP CS core network. This evaluation should be done in a case by case basis.

The main objective of this Technical Report is to provide a SIP-I based Nc interface that can suitably interwork with external SIP-I based signalling networks. These SIP-I based signalling networks are already prevalent in many fixed-line operator networks and transit networks today. The networks utilise the SIP profile that is defined by ITU-T Recommendation Q.1912.5 [16] Profile C which employs full ISUP encapsulation.

In order to ease interworking and improve convergence between fixed implementations and mobile implementations, the SIP-I based Nc interface shall be based on ITU-T Recommendation Q.1912.5 [16] Profile C.

5.2.2 Initial profile

Table 5.2.2.1 lists the references defined to be part of the Profile C of ITU-T Recommendation Q.1912.5 [16] that are applicable to the SIP-I based Nc interface. Additional elements of the 3GPP SIP-I profile are identified elsewhere within this TR.

It should be noted that some referenced RFCs provide additional procedures that are not applicable to the SIP-I based Nc interface (e.g., only ISUP MINE is required from RFC 3204 [54], QSIG is not within the scope of this 3GPP SIP-I profile). More explicit applicability will be provided by the SIP-I based Nc technical specification.

Table 5.2.2.1 Initial profile for SIP-I based Nc interface

Reference	ITU-T Profile C Status	SIP-I based Nc Status
RFC 2046 (November 1996) "Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types". [52]	Supported	Supported
RFC 2976 (October 2000): "The SIP INFO method". [53]	Supported	Supported
RFC 3204 (December 2001) "MIME media types for ISUP and QSIG Objects". [54]	Supported	Supported
RFC 3261 (June 2002): "SIP: Session Initiation Protocol". [30]	Supported	Supported
RFC 3262 (June 2002): "Reliability of provisional responses in Session Initiation Protocol (SIP)". [36]	Optional	Supported
RFC 3264 (June 2002): "An Offer/Answer Model with Session Description Protocol (SDP)". [27]	Supported	Supported
RFC 3311 (September 2002): "The Session Initiation Protocol (SIP) UPDATE method". [37]	Supported	Supported
RFC 3312 (October 2002): "Integration of resource management and Session Initiation Protocol (SIP)". [38]	Optional	Supported
RFC 3323 (November 2002): "A Privacy Mechanism for the Session Initiation Protocol (SIP)". [55]	Supported	Supported
RFC 3325 (November 2002): "Private Extensions to the Session Initiation Protocol (SIP) for Network Asserted Identity within Trusted Networks". [56]	Supported	Supported
RFC 3326 (December 2002): "The Reason Header Field for the Session Initiation Protocol (SIP)". [57]	Supported	Supported
RFC 3966 (December 2004): "The tel URI for Telephone Numbers". [31]	Supported Replaces RFC 2806	Supported
RFC 4566 (July 2006): "SDP: Session Description Protocol". [58]	Supported Replaces RFC 2327 [26]	Supported

The IETF has replaced or updated RFC 2806 with RFC 3966 [31] and RFC 2327 [27] with RFC 4566 [58] as shown in the table.

5.2.3 SIP-I profile options

Sub-clauses "Requirements on Nc interface" specify the minimum functionalities required within a closed 3GPP bearer independent circuit switched core network or for direct interoperation between 3GPP PLMNs.

5.2.3.1 Support for 100rel

5.2.3.1.1 Requirements on Nc Interface

Further to using SCTP to ensure a reliable transport service, IETF RFC 3262 [36] 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, see Clause 5.2.3.3), the support of 100rel as defined by IETF RFC 3262 [36] is mandatory for the 3GPP SIP-I profile on the Nc interface.

The following rules for negotiation support of 100rel shall be followed:

- A 3GPP node originating a SIP INVITE shall advertise its preference of provisional reliable responses via a SUPPORTED header containing the tag "100Rel".
- A 3GPP node (e.g. MSC-S) 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 3GPP node 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.

5.2.3.1.2 Requirements for interworking to external SIP-I networks

Some external networks may not include a REQUIRE header or SUPPORTED header containing tag "100rel" in its SIP messages. In Q.1912.5 support of 100rel is optional, however applications of SIP-I in external networks may mandate this support (e.g. NICC ND1017:2006/07 [71]). Three Alternatives exist currently if an external SIP-I network does not support 100rel and need to be decided for 3GPP networks (otherwise Clause 5.2.3.1.1 applies):

The following are several possible options for handling provisional responses. Further investigation is required to determine the final approach.

Alternative 1:

- A GMSC-S receiving a SIP INVITE request from an external network without a "100rel" tag in a REQUIRE or SUPPORTED header shall not include a REQUIRE header with tag "100rel" when sending a response in the range 101-199 and shall not send provisional responses reliably.

  However, the GMSC-S node, if sending a SIP INVITE request to a succeeding 3GPP node on the Nc interface, shall advertise its preference of provisional reliable responses via a SUPPORTED header containing the tag "100Rel" and respond to a provisional response containing the "100rel" tag with a PRACK request, as described in Clause 5.2.3.1.1.
- An MSC/IWF receiving a response in the range 101-199 without a "100rel" tag in a REQUIRE header shall not generate PRACKs. However the MSC/IWF shall return the response to the preceding node containing tag "100Rel".

Alternative 2:

- A GMSC-S receiving a SIP INVITE request without a "100rel" tag in a REQUIRE or SUPPORTED header shall not include a REQUIRE header with tag "100rel" when sending a response in the range 101-199 and shall not send provisional responses reliably.

  The GMSC-S shall forward the SIP INVITE request as received to the succeeding 3GPP node on Nc interface, i.e. with or without the tag "100Rel".

- An MSC/IWF receiving a response in the range 101-199 without a "100rel" tag in a REQUIRE header shall forward the response (without tag "100rel") on to the preceding node.

Alternative 3:

- A GMSC-S receiving a SIP INVITE request without a "100rel" tag in a REQUIRE or SUPPORTED header shall consider the INVITE request as erroneous and reject the INVITE request with a 421 Extension Required response.

- An MSC/IWF receiving a response in the range 101-199 with a REQUIRE header present without tag "100rel" shall consider the response as erroneous and reject the call accordingly.

5.2.3.2 Support for UPDATE method

5.2.3.2.1 Requirements on Nc interface

The UPDATE method as defined by IETF RFC 3311 [37] allows updating of session parameters (media streams, codecs) without modifying the dialog of a session. The UPDATE method is tightly associated with the use of preconditions to provide Continuity Testing which is a mandatory requirement.

The support of the UPDATE method shall be negotiated using the following rules:

- A 3GPP node originating a SIP INVITE request shall advertise its support of the UPDATE method via the ALLOW header listing the UPDATE method.

- A 3GPP node 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 as defined in IETF RFC 3311 [37], 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 3GPP node 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 as defined in IETF RFC 3311 [37].

5.2.3.2.2 Requirements for interworking to external SIP-I networks

In Q.1912.5 support of UPDATE is mandatory however the support of "Reliability of Provisional Responses" (IETF RFC 3262 [36]) is optional and thus signalling of UPDATE during early dialogues may not be possible. Applications of SIP-I in external networks may mandate this support (e.g. NICC ND1017:2006/07 [71]). Three Alternatives exist currently if an external SIP-I network does not signal indication of support of UPDATE method (otherwise Clause 5.2.3.2.1 applies):

The following are several possible options for handling of the UPDATE request. Further investigation is required to determine the final approach.

Alternative 1:

- A GMSC-S receiving a SIP INVITE from an external SIP-I network with the UPDATE method not included in the ALLOW header shall not generate the UPDATE method toward the external SIP-I network. However, it shall send SIP INVITE with ALLOW header listing the UPDATE method to the succeeding 3GPP node on the Nc interface.

- An MSC/IWF receiving a response to a SIP INVITE request shall check that an ALLOW header is present listing the UPDATE method. If the response contains an ALLOW header containing the UPDATE method, the GMSC-S is allowed to generate the UPDATE method towards the external network as defined in IETF RFC 3311 [37] for the purpose of session modification during early dialog. If the UPDATE method is not listed in the ALLOW header the GMSC-S shall not generate the UPDATE method towards the external SIP-I network. However the IWF/MSC shall return the response to the preceding 3GPP node containing an ALLOW header containing the UPDATE method.

- When the UPDATE method is not allowed on an interface during early dialog, changes to session parameters shall generally be sent using the re-INVITE method after the session has been established, although in some cases PRACK may be used.

Alternative 2:

- A GMSC-S receiving a SIP INVITE from an external SIP-I network which receives the ALLOW header without the UPDATE method shall then not generate the UPDATE method toward the external SIP-I network.
- The GMSC-S shall forward SIP INVITE request as received to the succeeding 3GPP node on the Nc interface, i.e. with or without the UPDATE method included in the ALLOW header.
- A MSC/IWF receiving a response to a SIP INVITE shall check that an ALLOW header is present listing the UPDATE method. If the response contains an ALLOW header containing the UPDATE method, the GMSC-S is allowed to generate the UPDATE method as defined in IETF RFC 3311 [37]. If UPDATE is not listed in the ALLOW header the GMSC-S shall not generate the UPDATE method towards the external SIP-I network.
- The MSC/IWF shall return the response to the preceding 3GPP node as received, i.e., with or without the UPDATE method included in the ALLOW header.
- When the UPDATE method is not allowed on an interface during early dialog, changes to session parameters shall generally be sent using the re-INVITE method after the session has been established, although in some cases PRACK may be used.

Alternative 3:

- A GMSC-S receiving a SIP INVITE from an external SIP-I network containing the ALLOW header without the UPDATE method shall consider the INVITE as erroneous and reject the INVITE request with a 421 Extension Required response.
- An MSC/IWF receiving a response with an ALLOW header without the UPDATE method the MSC/IWF shall consider the response as erroneous and reject the call accordingly.

5.2.3.3 Support for Preconditions

5.2.3.3.1 Requirements on Nc Interface

Support for preconditions as defined in IETF RFC 3312 [38] is optional as defined in Profile C of ITU-T Q.1912.5 [16]. The use of SIP Preconditions as described in IETF RFC 3312 [38] will allow a 3GPP node to progress the call forward before all originating side bearer resources have been allocated. This may occur for several reasons, including:

- Incoming ISUP IAM indicates COT on previous call leg or COT on this circuit
- Incoming BICC
- Originating RANAP is waiting for the network determined Selected Codec and Available Codec List.

The called subscriber shall not be alerted until preconditions have been satisfied.

To support these conditions, support for SIP Preconditions shall be mandatory within the 3GPP SIP-I Profile. Only the segmented procedures shall be used in order to allow each SIP end point to control its own precondition status.

Negotiating support of preconditions shall be done using the following rules:

- A 3GPP node originating a SIP initial INVITE request when preconditions have not been satisfied shall advertise its preference of preconditions via a SUPPORTED header containing the tag "precondition". The MSC-S shall encode preconditions in the SDP offer as specified in IETF RFC 3312 [38].
- When a 3GPP node receives a SIP initial INVITE request with the tag "precondition" in the SUPPORTED header, it shall include a REQUIRE or SUPPORTED header with tag "precondition" when sending a provisional 101 -199 response. The 3GPP node is then allowed to use preconditions as defined in IETF RFC 3312 [38].
- A 3GPP Node receiving a provisional 101 -199 response to a SIP initial INVITE request with tag "precondition" is then allowed to use preconditions as defined in IETF RFC 3312 [38].
- An MSC-S sending a SIP re-INVITE request or UPDATE request during a confirmed dialogue shall not include a REQUIRE or SUPPORTED header with tag "precondition".
- An MSC-S receiving a SIP re-INVITE request or UPDATE request during a confirmed dialog with a REQUIRE header containing the tag "precondition" shall reject re-INVITE request with a 420 'Bad Extension' final response.

5.2.3.3.2 Requirements for interworking to external SIP-I networks

In Q.1912.5 support of Preconditions is optional. Applications of SIP-I in external networks may mandate this support or even exclude it (e.g. NICC ND1017:2006/07 [71]). Three Alternatives exist currently if an external SIP-I network does not receive indication of support of Preconditions (otherwise Clause 5.2.3.3.1 applies):

The following are several possible options for handling of SIP Preconditions. Further investigation is required to determine the final approach.

Alternative 1:

- A GMSC-S receiving a SIP INVITE request without a "preconditions" tag in a REQUIRE or SUPPORTED header shall assume the preconditions have been satisfied. The GMSC-S shall not include a tag "precondition" when sending a response to the external SIP-I network.
- The GMSC shall either send the SIP initial INVITE request as received to the succeeding 3GPP Node i.e., without the tag "precondition" and appropriate SDP lines, or send the SIP initial INVITE request with the tag "preconditions" and with the indication that preconditions are met.
- An MSC/IWF receiving a provisional 101 -199 response to a SIP initial INVITE request without a REQUIRE or SUPPORTED header containing tag "precondition" shall continue the call without the use of preconditions. The MSC/IWF shall forward the response to the preceding 3GPP Node as received from the external SIP-I network (i.e., without the tag "precondition").

NOTE: Continuing the use of preconditions only between the originating MSC and the MSC/IWF has no useful function, the terminating network will continue to allow the call to proceed to answer, regardless of preconditions status. This option contradicts the general principle for alternative1 and may be considered actually part of alternative 2, rewording of this clause may be revised to reflect this during the conclusions phase.

A mechanism which prevents the alerting of the terminating device that does not support preconditions is FFS.

Alternative 2:

- A GMSC-S receiving a SIP INVITE request without a "preconditions" tag in a REQUIRE or SUPPORTED header shall behave as specified in alternative 1.
- An MSC/IWF receiving an initial INVITE request with a "preconditions" tag shall resolve the precondition within the 3GPP network before allowing the call to progress into the external SIP-I network. The initial INVITE request to the external network will not include a "preconditions" tag

Alternative 3:

- A GMSC-S receiving a SIP INVITE request without a "preconditions" tag in a REQUIRE or SUPPORTED header shall behave as specified in alternative 1
- An MSC/IWF receiving a a provisional 101 -199 response to a SIP initial INVITE without a REQUIRE or SUPPORTED header containing tag "precondition" when the initial INVITE request indicated that preconditions had not been satisfied shall consider the response as erroneous and reject the call accordingly.

5.2.3.4 Support for INVITE request without SDP

5.2.3.4.1 Requirements on Nc Interface

A 3GPP node may support the handling of re-INVITE request without SDP in sending and the receiving side to support optional features such as Bearer Redirection.

Scenarios shall be described in the TR to justify support in SIP-I based Nc of sending of re-INVITE requests without SDP so that the proposal can be completely assessed.

A 3GPP node originating a call shall always include SDP with the Supported Codec List in SIP Initial INVITE request to enable initial codec negotiation. A 3GPP node sending INVITE request without SDP is FFS.

5.2.1.4.2 Requirements for interworking to external SIP-I networks

Applications of SIP-I in external networks may mandate INVITE with SDP in all cases (e.g. NICC ND1017:2006/07 [71]). Three Alternatives exist currently if an external SIP-I network does signal INVITE without SDP (otherwise Clause 5.2.3.4.1 applies):

The following are several possible options for handling of INVITE request without SDP. Further investigation is required to determine the final approach.

Alternative 1:

- A GMSC-S, which receives an INVITE without SDP, shall create an own list of Supported Codecs containing all codecs supported by the GMSC-S (and its MGW) and generate an INVITE with its own SDP and send this to the succeeding 3GPP nodes.

Alternative 2:

- A GMSC-S or MSC/IWF, which receives an INVITE without SDP, shall forward the INVITE to the succeeding node without SDP. A 3GPP Node terminating the SIP-I on Nc shall create an own list of Supported Codecs containing all codecs supported by the 3GPP Node(and its MGW) and generate a response with its own SDP back to the preceding node.

Alternative 3:

- A GMSC-S receiving a SIP INVITE without SDP shall consider the INVITE as erroneous and reject the call accordingly.

5.2.3.5 Support for SDP with unspecified connection address

5.2.3.5.1 Requirements on Nc Interface

The support of an unspecified connection address in SDP using SIP-I based Nc is FFS.

5.2.3.5.2 Requirements for interworking with external SIP-I networks

In Q.1912.5 Profile C support of INVITE with unspecified connection address is implied by reference to offer/answer procedures as described in RFC 3264. However Q.1912.5 Profile C does not describe specific applications of this address. Applications of SIP-I in external networks may mandate INVITE with specified connection address in all cases (e.g. NICC ND1017:2006/07 [71]). Three Alternatives exist currently if an external SIP-I network does permit sending of SDP Offers with unspecified connection address (otherwise Clause 5.2.3.5.1 applies):

The following are several possible options for SDP with an unspecified address. Further investigation is required to determine the final approach.

Alternative 1:

- A GMSC-S, which receives an INVITE request with unspecified connection address shall seize a MGW and insert its own connection address in INVITE request to the succeeding 3GPP Node and in response back to the external SIP-I network.

Alternative 2:

- A GMSC-S, which receives an INVITE with unspecified connection address, shall forward the INVITE to the succeeding 3GPP node. The handling of the unspecified connection address at the terminating node is for further study.

Alternative 3:

- A GMSC-S receiving a SIP INVITE request with unspecified connection address shall consider the INVITE request as erroneous and reject the call accordingly.

5.2.3.6 SIP session continuity

The basic SIP protocol as defined by IETF RFC 3261 [30] does not include a "keep alive" mechanism. As such, it is possible that one end of a session may fail and be unable to signal the release of the session.

One possible scenario where this may occur is in the cases where an internal fault on a remote node results in the call instance being lost on the remote node. This would result in no further signalling from the remote node associated with that call instance. The local node would have no indication from the remote node should that party release the call.

The SIP Session Timer as described in IETF RFC 4028 [51] provides a means to determine whether a SIP session is still active by attempting to perform a session refresh. 3GPP nodes can take advantage of this mechanism to know when resources may be released if one end of the session fails.

The procedures negotiate the rate at which the session refresh occurs. The procedures are compatible and still operational should the far end or other SIP network not support the Session Timer procedures.

The following alternatives to check aliveness of the session were also discussed but none was considered appropriate for the reasons explained below:

1. Use of the SIP OPTIONS message:

   - The SIP OPTIONS message is more generally used outside dialogs to check the connectivity between SIP nodes, not to check the aliveness of specific sessions.

   - Use of the SIP OPTIONS message would not allow negotiating end to end an agreed "keep-alive" timer. This would therefore force intermediate nodes to generate OPTIONS messages on their own too, in addition to the OPTIONS messages sent end to end between the UAs. This would also possibly lead UAs to use inappropriate timers for checking aliveness of sessions (e.g. too short timers).

   - The SIP Session Timer is the standard SIP "keep alive" mechanism specified by IETF, so there's only benefit to stick to IETF recommendations to remain compatible with what will be also supported by external SIP/SIP-I networks.

2. Use of periodic RTCP packets:

   - RTCP addresses media level control, but does not address signalling control.

   - This would not allow intermediate nodes without MGW to check aliveness of sessions.

   - This would mandate support of RTCP protocol to check aliveness of sessions.

3. Use of ISUP timers:

   - None allows detecting the loss of a particular call instance once the call is in a stable phase (call answered).

4. Use of SCTP transport association status:

   - This would not allow checking the aliveness of particular sessions (the SCTP transport association may remain active even after the loss of a particular call instance).

   - This would not allow to check aliveness of a particular session in a remote entity, but just to check aliveness of the transport association with the next hop node.

   - This would not be usable would other transports be required towards external SIP-I networks not supporting SCTP.

Therefore the SIP Session Timer procedure may be supported as an option on SIP-I based Nc.

5.3 Transport

SIP is a transport-independent protocol and as such can run over any reliable or unreliable message or stream transport. Traditionally SIP clients use TCP and UDP to connect to SIP servers and SIP endpoints. However, IETF RFC 4168 [23] specifies a mechanism for usage of SCTP as the transport mechanism between SIP entities. SCTP can provide added reliability for SIP signalling and has advantages over both TCP and UDP as specified in IETF RFC 4168 [23] (e.g. Fast Retransmit, Congestion Control, Multihoming, etc.).

Furthermore, SCTP is also a valid transport protocol within the IMS SIP profile as specified in 3GPP TS 24.229 [24].

SCTP is specified in IETF RFC 2960 [60] and the 32 bit CRC checksum mechanism is specified by IETF RFC 3309 [61]. This SCTP transport is consistent with that used within the current protocol architecture as specified in 3GPP TS 29.202 [59].

For the added reliability and compatibility with existing SIP profiles, SCTP shall be the transport protocol for a SIP-I based Nc interface as per IETF RFC 4168 [23].

For improved resilience, once established, SCTP associations should be re-used with all SIP messages.

5.4 Impacts on the Bearer Plane

5.4.1 Bearer selection for SIP-I based Nc

SIP-I as currently defined supports a specific set of codecs and associated with that, specific forms of framing for those codecs. These are mostly based on the expectation that the codecs used in the bearer plane are transported directly on to IETF defined transport and framing. However, the current Nb interface is defined to allow for the continuation of the Iu User Plane Framing Protocol into the Core Network with minimal modification, as described in 3GPP TS 29.414 [12] and 3GPP TS 29.415 [13].

Currently there is no defined way for a SIP-I based call control interface to support the establishment of bearers supporting the Nb User Plane Framing Protocol. Therefore, two choices exist:

1. define extensions for SIP-I to allow the protocol to support Nb User Plane Framing Protocol.
2. use existing IETF specified framing protocols for the transport of CS traffic in the bearer plane.

Defining extensions to SIP-I would be both a lengthy process (involving at least the IETF and possible ITU-T) and also would seem to be defeating the object of using SIP-I. The move towards a SIP-I based Nc interface is driven in part by the potential to simplify the interworking of the interconnect between networks. This interconnection will require that the bearer plane is in some cases modified to a widely adopted framing protocol at the network edge anyway, so doing this translation at the MGW where the RAN meets the CN offers no disadvantage.

In addition, the Mb interface for IMS (as defined in 3GPP TS 29.163 [25]) is already defined as being based on RTP framing and so, moving to the use of RTP on the bearer plane when SIP-I is used in the control plane will align the Nb and Mb interfaces for the transport of CS traffic.

Moving to RTP as the transport protocol will have some impact on the functioning of the Mc interface (see 3GPP TS 29.232 [10]), but this impact can be set against potential reuse of the procedures and properties from the Mn interface profile as defined in 3GPP TS 29.332 [20], which is used to control RTP-based bearers in the IMS domain.

Therefore the bearer for a network supporting a SIP-I based Nc interface shall be an RTP-based bearer, and shall align with the Mb interface 3GPP TS 29.163 [25].

5.4.2 Bearer transport with multiplexing

SIP-I based Nc shall be able to provide at least the same bandwidth efficiency as BICC-based Nc.

An optional transport format is specified for BICC-based CSCN for the Nb interface and IP transport that allows transporting several RTP/NbFP/codec payload PDUs of different user plane connections within one packet, with the corresponding backward compatible signalling extensions required to negotiate the use of this transport option.

In SIP-I based Nc, the /IP/UDP/RTP header overhead (40 byte for IPv4, 60 byte for IPv6) and the layer 2 header/trailer header overhead (e.g. 38 bytes for Ethernet v2, 7 bytes for POS) are still much larger than the transported payload codec (e.g. 31 byte for AMR 12.2 kHz and 5 byte for SID frames). It is much desirable to reduce the /IP/UDP/RTP header and the layer 2 header/trailer overheads to save bandwidth and ensure a bandwidth efficiency equivalent to BICC-based Nc.

An optional transport format shall therefore also be specified in in SIP-I based Nc to allow multiplexing several codec payload PDUs of different bearers within one IP packet sent between MGWs over the Nb interface, and allow as an option compressing RTP headers, in a similar way as what has been defined for BICC-based CSCN:

- UDP port multiplexing should be re-used as specified for BICC-based CSCN ;

- RTP header compression should rely on the principles defined in section 5.1.2.2 (RTP Header Reduction for VoIP transport with RTP framing) of 3GPP TR 29.814 [68] ;
- Multiplex negotiation should rely on the same principles as those defined for BICC-based CSCN (based on RTCP signalling).

The detailed procedures will be defined during the stage 2 / stage 3 specifications.

5.5 Support of Media Description

Description of the media within SIP-I is provided through the use of SDP, as described in IETF RFC 4566 [hh] and further described for SIP-I in ITU-T recommendation Q.1912.5 [16]. For the purpose of a SIP-I based Nc interface, these mechanisms are re-used.

In the current BICN architecture, the BICC based Nc interface utilises IPBCP to exchange the IP bearer information between peer MGW's as defined in 3GPP TS 29.414. IPBCP is tunnelled over the Mc interface and Nc interface in the BICN architecture. In a network utilising a SIP-I based Nc, the support of SDP provides this equivalent functionality to the support of IPBCP in the existing BICC based Nc interface between the call server functions. With SIP-I, two approaches to the support of media description on the Mc interface can be considered:

- Maintain the Mc interface through interworking the SDP from the SIP-I Nc interface into the IPBCP container in the H.248 commands on the Mc interface.
- Introduce new properties and procedures to transport the SDP information in the H.248 commands on the Mc interface.

The first of these options minimises the impact on the existing Mc interface, but it is clear that IPBCP would provide unnecessary overhead and require an interwork function in the MSC-S to support the encapsulation of the SDP within IPBCP for transport across Mc. The second alternative of transporting the SDP information within H.248 is not currently supported in Mc interface specifications, but is very much aligned with the Mn interface (see 3GPP TS 29.332 [20]). In fact, it is likely that much of the specification of the properties and procedures defined in the Mn interface could be reused in conjunction with a SIP-I based Nc interface.

Therefore, support of transporting the SDP information directly within the H.248 commands, and the re-use of existing properties and procedures defined in the Mn interface specification is recommended where possible.

5.6 Bearer Establishment Models

Currently the BICN architecture supports four models of bearer establishment models for an IP bearer as defined in 3GPP TS 23.205 [2]. These are listed below:

- Fast Forward/Backward: Tunnel Data is transferred by the originating MSC-S as part of the IAM message to the terminating MSC-S. The Tunnel Up procedure is transferred from the originating MGW to the originating MSC-S, prior to sending the IAM, in order to retrieve the tunnel data. The terminating MSC-S exchanges Tunnel Up/Down procedures with the terminating MGW and the terminating MSC-S transfers this tunnel data to the originating MSC-S in a Tunnel Information message. The originating MSC-S uses the Tunnel Down procedure to the originating MGW to supply the received tunnel data. This procedure does not permit optimised MGW selection, i.e. the originating node must select a MGW before sending the IAM.
- Delayed Forward: Tunnel Data is transferred by the originating MSC-S in the Tunnel information message following the Bearer Information message received from the terminating MSC-S. The Tunnel Up procedure is transferred from the originating MGW to the originating MSC-S to retrieve the tunnel data to be added to the Tunnel Information message transferred from the originating MSC-S to the terminating MSC-S. The terminating MSC-S exchanges Tunnel Up/Down procedures with the terminating MGW and the terminating MSC-S transfers this tunnel data to the originating MSC-S in a Tunnel Information message. The originating MSC-S uses the Tunnel Down procedure to the originating MGW to supply the received tunnel data. This procedure does permit optimised MGW selection in the originating call side.
- Delayed Backward: Tunnel Data is transferred by the terminating MSC-S following the IAM. The Tunnel Up procedure is transferred from the terminating MGW to the terminating MSC-S to retrieve the tunnel data to be added to the Tunnel Information message which in turn is transferred by the terminating MSC-S to the originating MSC-S. The originating MSC-S exchanges Tunnel Up/Down procedures with the originating MGW

and the originating MSC-S transfers this tunnel data to the terminating MSC-S in a Tunnel Information message. The terminating MSC-S uses the Tunnel Down procedure to the terminating MGW to supply the received tunnel data. This procedure permits optimised MGW selection in the terminating call side.

The 'Offer/Answer' model as specified in IETF RFC 3264 [27] has the functionality to be able to provide an equivalent of the Fast Forward bearer establishment model described above within a network utilising a SIP-I based Nc interface, as the media information is included in the SDP of the initial offer.

The Delayed Backward establishment model can also be supported via using the unspecified connection address in the initial SDP offer. In this case always a full second SDP offer/answer exchange is needed to convey bearer parameters between the endpoints.

In addition, it may be possible to initiate a session without the SDP in the initial SIP INVITE. It is for further study whether this ability will be allowed between 3GPP entities, however for interworking with external SIP-I networks the receipt of such a SIP INVITE shall be supported. In such a case, the terminating MSC-S may then initiate the initial offer and include the media information within the SDP.

Therefore the proposal is that a SIP-I based Nc interface shall support the Fast Forward bearer establishment model. The support of the Delayed Backward bearer establishment model is FFS. It is not proposed to support any other bearer establishment model.

5.7 Codec Negotiation

5.7.1 Codec Negotiation Offer/Answer Model

The support of Q.1912.5 has associated with it support for the 'Offer/Answer' model as specified in IETF RFC 3264 [27]. The Offer/Answer model is capable of exchanging SDP for the purposes of session establishment and can be used to support the speech codec negotiation that is required to support the OoBTC functionality. Codecs identified in IETF RFC 3267 [18], IETF RFC 3551 [28] and IETF RFC 3555 [29] may be supported for OoBTC. As a minimum, the mandatory speech codecs required to be supported are listed in 3GPP TS 23.153 [17].

If the answer includes multiple speech codecs from the initial offer, then all of these codecs are available for immediate use unless the offerer sends a second offer to limit the list of active codecs. It is beneficial for both endpoints in a 3GPP SIP-I network to be aware of the Available Codecs List, which can only be signalled using existing IETF procedures by including multiple speech codecs in the answer. To avoid a second offer/answer exchange, a 3GPP-specific SDP extension can be used to allow the inclusion of the Available Codecs List within the SDP answer in such a way that the speech codecs in the Available Codecs List are clearly identified as unavailable for use without an additional offer/answer exchange.

IETF RFC 3264 [27] states that the answerer may list the codecs in their desired order of preference, but it is recommended that unless there is a specific reason, the answerer lists the codecs in the same relative order they were present in the offer. However, the support of the codec negotiation process used for TrFO provides sufficient justification for the reordering of the speech codec information in the offer/answer mechanism.

NOTE: It is only the speech codecs that are considered in this section as it is the speech codec negotiation and support of multiple speech codecs simultaneously that is of issue. Other SDP "codecs" such as RTP Tel Event or Silence Frame payload may be supported simultaneously with other SDP "codecs". Where "codec" is mentioned in this Clause it is meant Speech Codec unless explicitly stated otherwise.

For the codec negotiation over a SIP-I based Nc interface, two models are considered:

1. An initial offer includes the codecs supported by the offerer in a descending order of preference, with the answer including the codecs from that list that are also supported by the answerer in the same order. This scenario is problematic if the answerer includes speech codecs in the answer that the answerer cannot support simultaneously. In order to resolve this, a second offer must be made identifying a single selected codec for the call/session that is being established however the SDP offer/answer procedures do not mandate that the offerer initiate a second offer/answer exchange. See Figure 5.7.1.1.

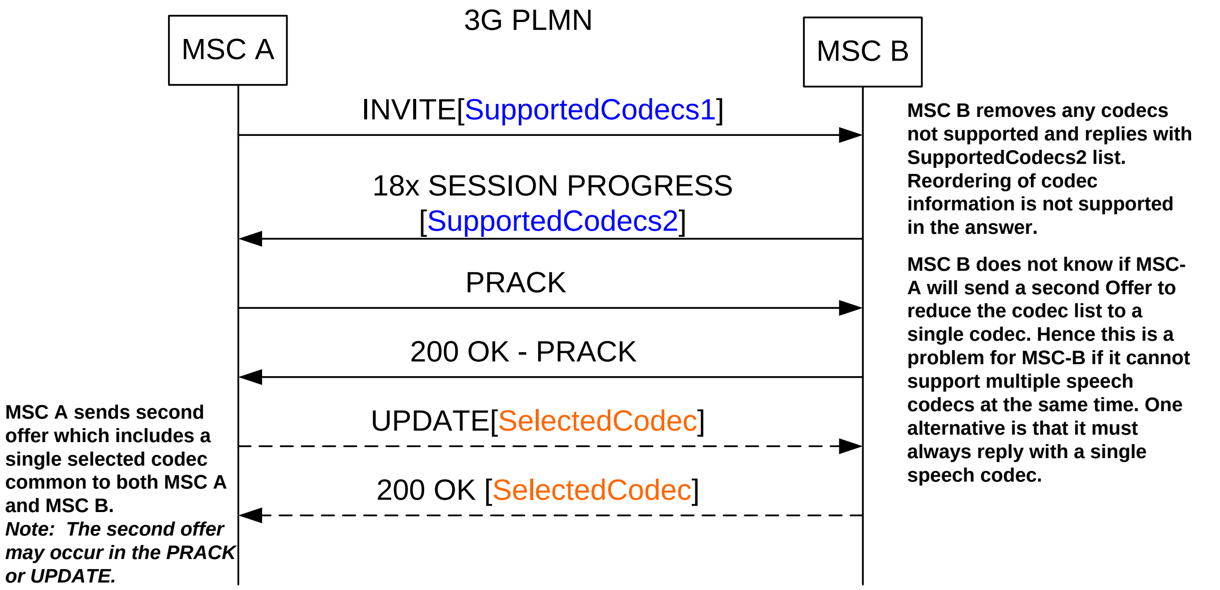

Figure 5.7.1.1: Codec Negotiation and Call Establishment with two Offer/Answer Method

2. An offer includes the codecs supported by the offerer in a descending order of preference, with the answer including the codecs from that list that are also supported by the answerer in a descending order of preference as determined by the answerer with the selected codec moved to the top of the list. In order to identify this codec negotiation as being in accordance to 3GPP specifications a 3G SDP extension is included in the SDP in both the Offer and the Answer. See Figure 5.7.1.2 shows and example of this proposal.

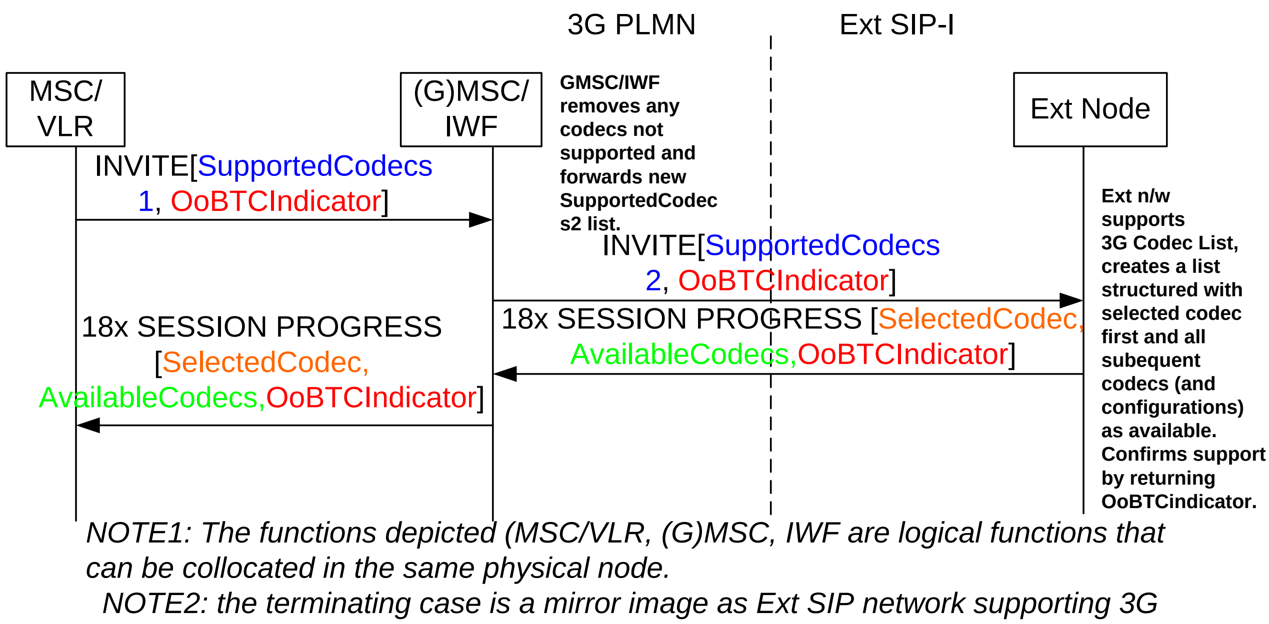

Figure 5.7.1.2: Codec Negotiation and Call Establishment with single Offer/Answer Method

Therefore, in Option 2 the codec negotiation mechanism on the Nc interface shall use the offer/answer mechanism with the following clarifications:

- A new 3GPP SDP extension will be defined that indicates that the offerer supports the 3GPP codec negotiation as described in this clause.

- A new 3GPP SDP extension will be defined that indicates that the answerer supports the 3GPP codec negotiation as described in this clause. The answer will contain a selected codec and additional "available codecs". The "available codecs" are clearly understood to be unavailable for use by either endpoint during the session without another offer/answer exchange
- If the offer includes the 3GPP SDP extension, then if the answerer supports the 3GPP codec negotiation as well the answer shall reply with the new 3GPP SDP extension as described above.
- The Selected Codec shall include all parameters and configuration information such as mode set in an exact and unambiguous way.
- The Codec entry for AMR Codec Types may have several additional parameters. These parameters may specify support of alternative AMR configurations in the Supported Codec List and in the Available Codec List, but these parameters shall specify exactly one AMR Configuration for the Selected Codec.
- The Available Codec List shall be ordered and structured in the same way as the Supported Codec List, but in priority of the Answerer.

  NOTE: This Available Codec List permits the Offerer to select an optimal new local access codec during a handover to another (radio) access type that may support other codecs than were originally offered to the Answerer. The Offerer can thus optimise the codec selection at the time of handover rather than having to select a codec that is not optimum and then try to modify this after a subsequent codec negotiation with the Answerer. Codec modification toward the terminal and RAN should be minimized.

- The SDP answer shall include the codecs from the received list that are also supported by the answerer in a descending order of preference as determined by the answerer with the Selected Codec moved to the top of the list. The remainder of the received list of codecs shall contain the Available Codecs List.
- The Answerer may also return additional codecs or configurations that it supports, but which were not in the initial Supported Codec List of the Offerer.
- An additional SDP offer/answer (codec) negotiation may be required whenever there is a change in the bearer configuration due to, e.g., handover or feature invocation.
- Mid-call codec negotiation shall use the SIP UPDATE method during an early dialog and may use either the SIP UPDATE method or the re-INVITE method during an established dialog, although IETF RFC 3311 recommends the use of the re-INVITE method in this case.

5.7.1.1 Sub-Proposals for Codec Negotiation Option 2

5.7.1.1.1 Option 2 Sub-Proposal 1

One proposal under option 2 is to mandate the above procedures within 3GPP Nc interface, i.e. between 3GPP logical nodes (as given by the example Figure 5.7.1.2).

In this proposal the logical Interworking Function (IWF) shall perform the following procedures towards an external network:

- If the Answerer includes a list of codecs and does not include the "OoBTCIndicator" from an external network and the Gateway Server is involved in the codec negotiation then it shall select the most suitable codec from the list and return this as the first codec in the list to the preceding node along with the "OoBTCIndicator". The Gateway node shall in addition send a second Offer to the external network with the single selected codec, see example Figure 5.7.1.1.1.1. It should be noted that the external network may not support PRACK and that IETF codec list can be received in 200 OK INVITE rather then 18x and if PRACK is not supported, even if the Codec list is received initially in 18x unreliably, GMSC has to wait 200 OK INVITE with the same Codec list before initiating second offer.

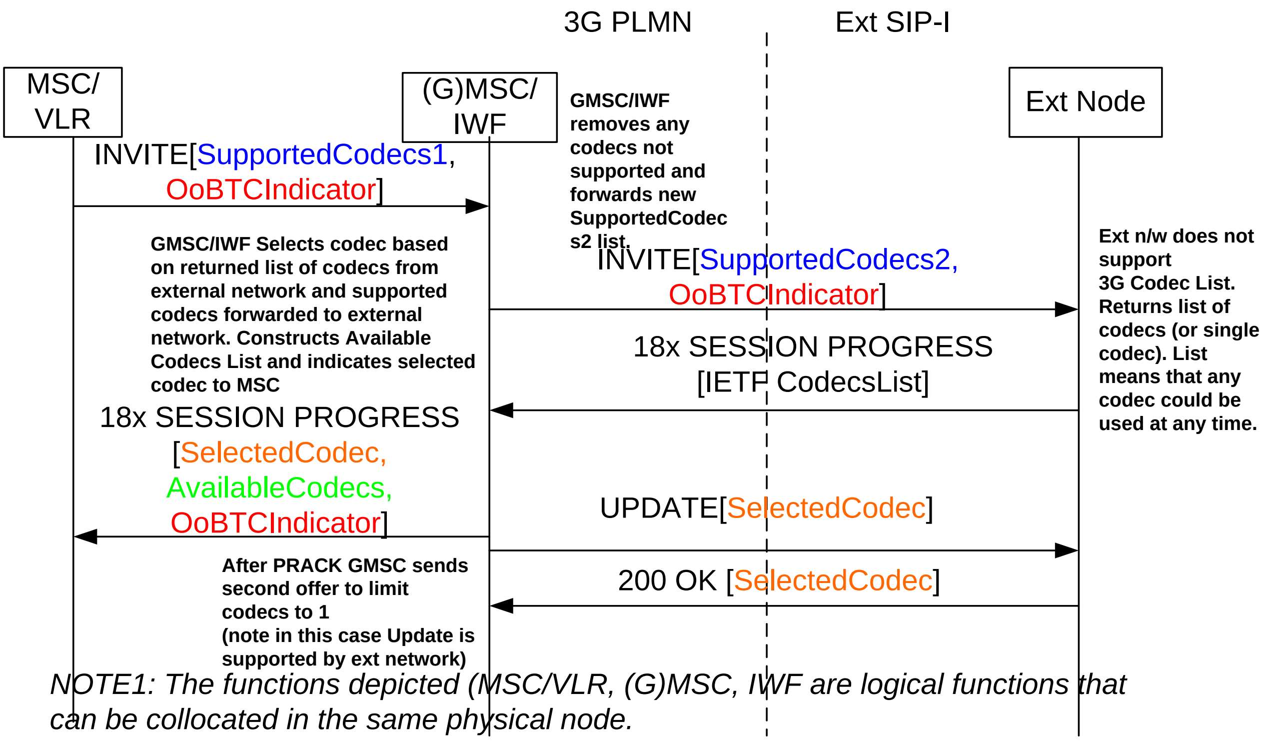

Figure 5.7.1.1.1.1: Codec Negotiation to external Non-3G SIP-I network

- If the offer does not include an indication of support for the "OoBTCIndicator" from an external network and the Gateway Server is involved in the codec negotiation it shall remove any unsupported codecs from the list and forward to the succeeding node including the "OoBTCIndicator". On receipt of the Answer from the succeeding node the Gateway Server shall return only the selected codec to the external non-3G network. See Figure 5.7.1.1.1.2.

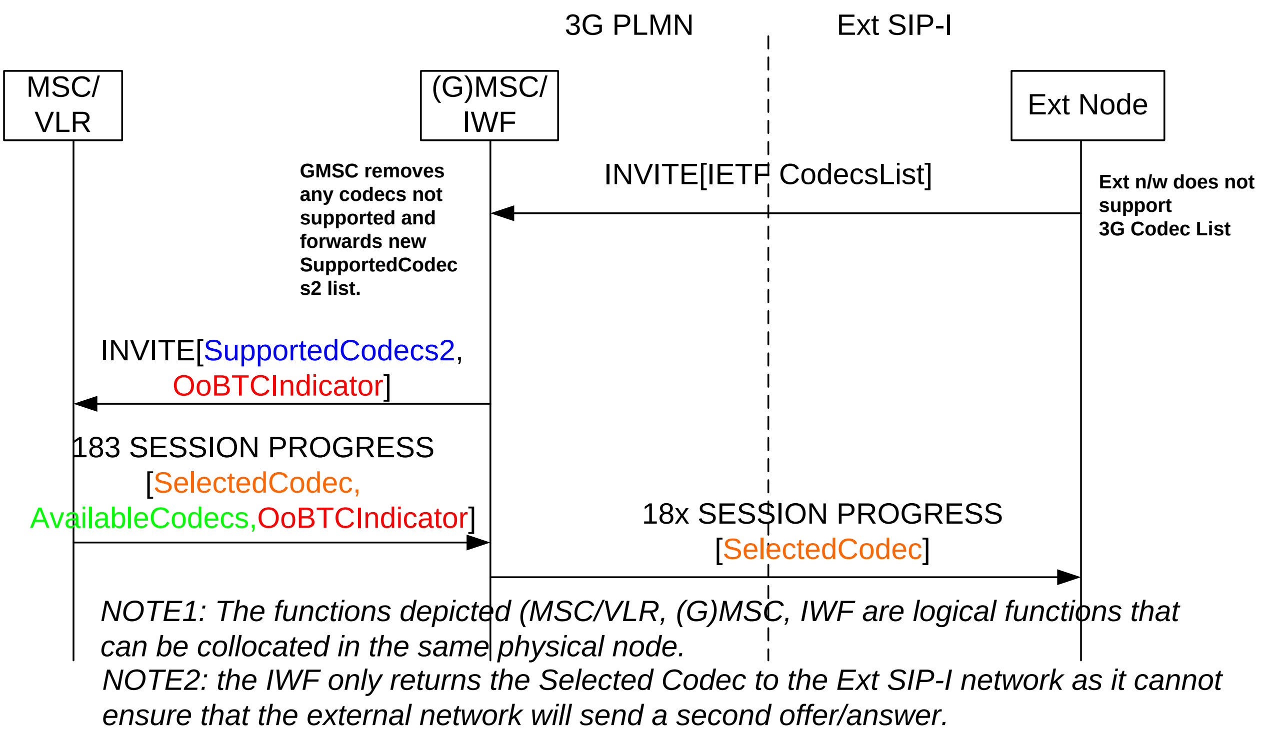

Figure 5.7.1.1.1.2: Terminating Codec Negotiation from Non-3G external SIP-I Network

- If the offer includes an indication of support for the "OoBTCIndicator", the SDP answer shall include the codecs from the received list that are also supported by the answerer in a descending order of preference as determined by the answerer with the Selected Codec moved to the top of the list and shall include the "OoBTCIndicator" in the Answer. The remainder of the received list of codecs shall contain the Available Codecs List.
- The Selected Codec shall include all parameters and configuration information such as mode set in an exact and unambiguous way.
- The Available Codecs List may contain alternative configurations of the Selected Codec.
- The Answerer may also return additional codecs or configurations that it supports, but which were not in the initial Supported Codec List of the Offerer.
- The Available Codec List shall be ordered and structured in the same way as the Supported Codec List, but in priority of the Answerer.

NOTE: This Available Codec List permits the Offerer to select an optimal new local access codec during a handover to another (radio) access type that may support other codecs than were originally offered to the Answerer. The Offerer can thus optimise the codec selection at the time of handover rather than having to select a codec that is not optimum and then try to modify this after a subsequent codec negotiation with the Answerer. Codec modification toward the terminal and RAN should be minimized.

5.7.1.1.2 Option 2 Sub-Proposal 2

In a second proposal within option 2 the support of the 3GPP codec negotiation procedures at the answering node are optional and thus the following procedures apply to 3GPP logical nodes MSC/VLR:

- If the Answerer receives an Offer without the 3G indicator then it shall assume a non-3G network and shall then include only those speech codecs in the answer that it can support simultaneously. The 3G network shall not send back in the Answer multiple codecs as it cannot ensure that the Non-3G network will return a subsequent Offer to limit the codecs to a single selected codec.
- If the Offerer receives the Answer without the 3G indicator then it shall assume a non-3G network. If the offerer selects not to support the combination of codecs in an SDP Answer, the offerer shall immediately send a subsequent Offer that contains an abbreviated list of codecs from the previous SDP.
- If the answer contains a list of codecs that cannot be supported simultaneously by the offerer, the offerer shall initiate a second offer/answer negotiation to limit the list to those that are supported.

The following figures provide several example sequence diagrams to illustrate possible offer/answer exchanges that can occur within a 3GPP network or between a 3GPP network and an external SIP-I network. These examples show various combinations of where the offerer or answerer either support or do not support the 3GPP SDP extension indicator. Additional examples are also included to show scenarios where a transit node is included in the path, either with or with a media gateway as well as scenarios where the transit exchange does not support the 3GPP SDP extension indicator and so provides optional transcoding. In these flows, the external SIP-I network represents a SIP-I network that does not support this extension. Should the external network support the extension, then the examples representing MSCs that support the extension would apply (with the MSC replaced by the external node).

Figure 5.7.1.1.2.1 shows a typical offer/answer exchange within a 3GPP network. If the offering exchange supports the 3GPP SDP extension it indicates so by including the new 3GPP SDP extension indicator in the initial offer. If the answering exchange supports the 3GPP SDP extension, it will provide in the SDP answer the selected codec and any additional available codec(s) as well as the new 3GPP SDP extension indicator. If the answerer does not support the 3GPP SDP extension, the answer will contain a list of codec(s).

Upon receiving the SDP answer, a non-3GPP answer format is received and the offerer is not able to simultaneously support all the codecs provided in the SDP answer, a new offer will be made to reduce the codec list to only those codecs that can be supported simultaneously. While the figure shows this new offer exchange using the UPDATE method, it can also be performed using the PRACK method. At this point, there is little value in including the 3GPP SDP extension indicator in the subsequent offer as the answerer has already demonstrated that it is not recognised.

NOTE: The term "Codec Info" in the SDP answer is used to represent the returned codec list. The exact format of this list, when the new 3GPP SDP extension is used, is for FFS.

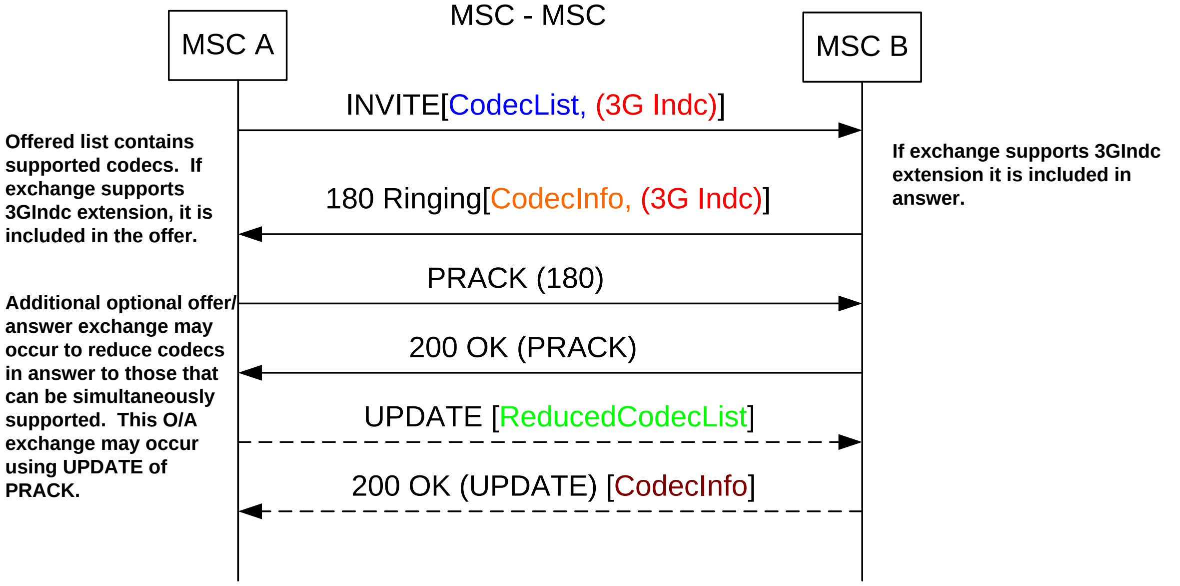

Figure 5.7.1.1.2.1 MSC to MSC Offer/Answer

Figure 5.7.1.1.2.2 shows a scenario similar to Figure 5.7.1.1.2.1 except that the answerer is in an external SIP-I network. In this case, this external node does not support the 3GPP SDP extension so the SDP answer is formatted following standard IETF rules and the 3GPP SDP indicator is not returned. If the offerer is not able to simultaneously support all the codecs provided in the answer, a new offer will be made to reduce the codec list to only those codecs that can be supported simultaneously.

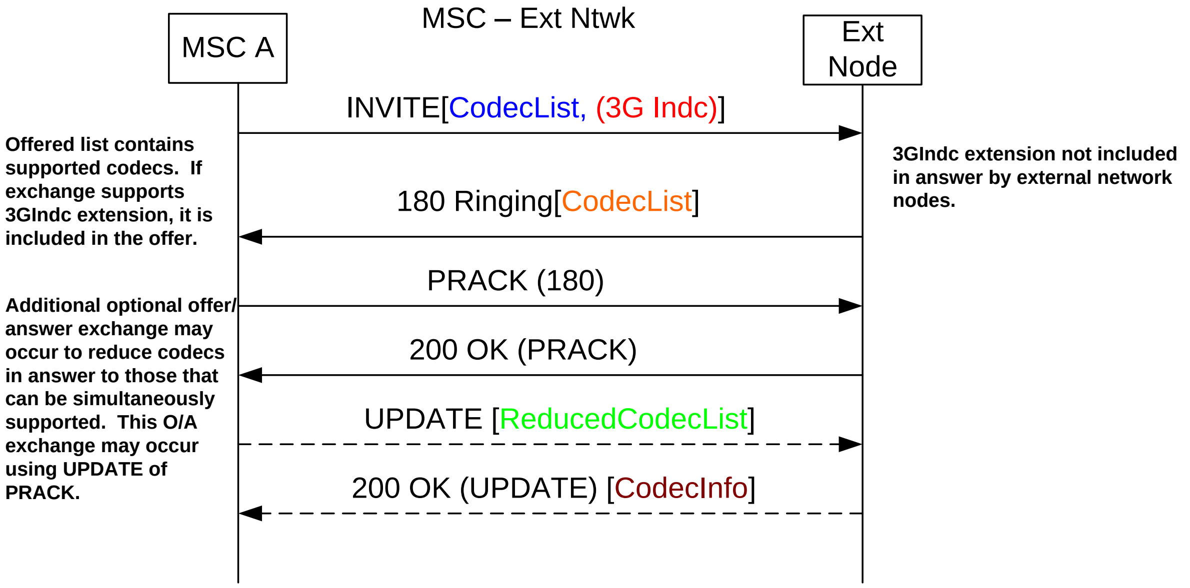

Figure 5.7.1.1.2.2 MSC to External SIP-I Network Offer/Answer

5.7.1.1.3 Comparison of Codec Negotiation Sub-Proposals

5.7.1.1.3.1 Sub-Proposal 1 - Mandatory support of 3GPP OoBTC Indicator within SIP-I over Nc in the Offer and Answer

- Emulation of the BICC codec negotiation procedures providing full compliance to 3GPP TS 23.153.
- A common solution for Codec Selection Nc interface using BICC and SIP-I within 3GPP networks
- Interworking to external networks is kept at the network border to provide interworking between the 3GPP and non-3GPP SIP-I profiles.
- It permits evolution of "OoBTC Indicator" into external networks (IETF draft is required), by using the procedures of Sub-Proposal 2 when interoperating with the external network.

5.7.1.1.3.2 Sub-Proposal 2 - Optional support of 3GPP OoBTC Indicator within SIP-I over Nc in the Offer and Answer

- A standard SIP-I codec negotiation method would be introduced on the Nc with a negotiable 3GPP extension (the OoBTC Indicator), thus allowing interworking with external SIP-I implementations.
- Only one SIP-I profile is needed within the 3GPP PLMN and when interconnecting to external SIP-I networks, thus no separate interworking is required for codec negotiation.
- Emulation of the BICC codec negotiation procedures in 3GPP TS 23.153 in scenarios only when negotiated using the optional OoBTC Indicator.
- When endpoints do not negotiate the use of the OoBTC Indicator, if the Answerer returns multiple codecs, the 3GPP Offerer is required to send a subsequent Offer/Answer to reduce the codec list to a single codec.

5.7.1.2 Conclusion of Codec Negotiation Offer/Answer Model

5.7.1.2.1 General

It is clear from Clause 5.7.1.1.3 that both Sub-proposals have advantages and disadvantages. Therefore a combination of the two sub-proposals is proposed to be used for the codec negotiation mechanism for a SIP-I based Nc Interface. The functionality is described in the following sections for both the 3GPP Node Originating and Terminating cases.

5.7.1.2.2 3GPP Node Originating SDP Offer

- An MSC-S initiating an offer shall include the OoBTC Indicator in the offer.
- If the offering MSC-S receives an answer without the OoBTC Indicator, the codec list shall be interpreted in accordance with the IETF codec rules. If the answer contains multiple codecs, the MSC-S initiates a second offer with the selected codec and may include the OoBTC Indicator or leave it out.
- If the offering MSC-S receives an answer with the OoBTC Indicator, then the codec list will contain the 3GPP Selected Codec and Available Codec List and shall be interpreted according to the 3GPP procedures.

5.7.1.2.3 3GPP Node Terminating SDP Offer

- If the 3GPP MSC-S terminating the codec negotiation receives an offer with the OoBTC Indicator, it shall include the OoBTC Indicator in the Answer. The returned codec list shall be formatted as a 3GPP Selected Codec List and Available Codec List.
- If the 3GPP MSC-S terminating the codec negotiation receives an offer without the OoBTC Indicator, the codec list shall be interpreted in accordance with the IETF codec rules and the MSC-S shall initiate an answer with a single codec. It may include the OoBTC Indicator in the Answer or leave it out.

5.7.1.2.4 Intermediate Node Receiving SDP Offer

- A 3GPP intermediate node receiving an offer with the OoBTC Indicator shall forward the OoBTC Indicator in the Offer to the succeeding node.
- A 3GPP intermediate node receiving an offer without the OoBTC Indicator shall behave according to two options dependent on implementation option:

  1. The 3GPP intermediate node may forward the IETF codec list to the succeeding node without the OoBTC Indicator.
  2. The 3GPP intermediate node may include the OoBTC indicator in the offer it sends to the succeeding node.

  NOTE: The intermediate node may forward an INVITE for a call that was initiated by the external network (External NW -> intermediate node(s) -> external NW), in which case the offer received from the preceding node may not contain the OoBTC Indicator.

5.7.1.2.5 3GPP Intermediate Node Receiving SDP Answer

- A 3GPP intermediate node receiving an answer with the OoBTC Indicator shall behave according to the presence of the OoBTC Indicator in the initial offer received from the preceding node.

  1. If the Initial Offer included the OoBTC Indicator, the 3GPP intermediate node shall forward the Answer with the OoBTC Indicator to the preceding node. The codec list shall be formatted as a 3GPP Selected Codec and Available Codec List.
  2. If the Initial Offer did not include the OoBTC Indicator, the 3GPP intermediate node shall forward the Answer without the OoBTC Indicator to the preceding node. The answer shall contain a single codec (mapped from the 3GPP Selected Codec).

- A 3GPP intermediate node receiving the answer with multiple codecs and without the OoBTC Indicator shall behave according to the three options below, dependent on implementation option:

  1. The 3GPP intermediate node may forward the IETF codec list to the preceding node, regardless of whether the preceding node included the OoBTC Indicator in the offer. The answer shall not contain the OoBTC Indicator.

     NOTE: This may be permitted when the intermediate node can support multiple speech codecs during a given session; if this is not the case then option 3 shall be performed.

  2. If the initial offer received by the intermediate node contained the OoBTC Indicator, the 3GPP intermediate node may map the IETF codec list into a 3GPP Selected Codec and Available Codec List and forward this to the preceding node with the OoBTC Indicator. This exchange concludes the offer/answer from the perspective of the preceding node. If the answer contains multiple codecs, the 3GPP intermediate node shall initiate a second offer toward the succeeding node with a single codec (same as the 3GPP Selected Codec) without the OoBTC Indicator.
  3. If the initial offer received by the intermediate node did not contain the OoBTC Indicator the 3GPP intermediate node may signal back to the preceding node a single codec (it shall select the most appropriate codec from the list of received codecs). This exchange concludes the offer/answer from the perspective of the preceding node. The 3GPP intermediate node shall initiate a second offer toward the succeeding node with a single codec (same as the 3GPP Selected Codec) without the OoBTC Indicator.

5.7.1.2.6 Semantics of 3GPP OoBTC Indicator

After the 3GPP OoBTC Indicator has been negotiated, i.e. the 3GPP OoBTC indicator has been included in both SDP offer and corresponding SDP answer the following rules apply for both offerer and answerer:

- A change from the Selected Codec in the answer to a codec within the Available Codec List (ACL) is only permitted using a new SDP offer-answer exchange to re-negotiate the Selected Codec. An "Inband" switching by sending the other codec with corresponding RTP payload type is not permitted, and no resources for these codecs need to be reserved e.g. at a MGW.

- Codecs in the Available Codec List indicate codecs that are supported. This information can be used by an MSC to decide if a change of the Selected Codec to some other codec using a new SDP offer-answer exchange will be attempted.
- A change from the Selected Codec in the answer to an auxiliary codec of "Miscellaneous" codec type within the answer, i.e. the telephone-event DTMF codec or the comfort noise codec (RFC3389), and vice versa is permitted without new SDP offer-answer exchange by "Inband" switching, i.e. by simply sending the other codec with corresponding RTP payload type. However, a switch between the comfort noise codec and the Selected Codec is only permitted if the comfort noise codec (RFC3389).is applicable for he selected codec. For instance, AMR does contain an internal comfort noise mode and is not used in combination with the comfort noise codec (RFC3389). Resources for a possible telephone-event DTMF codec within the answer shall therefore be reserved. Resources for a possible comfort noise codec (RFC3389) within the answer codec shall only be reserved if the comfort noise codec (RFC3389) is applicable for the selected codec.

NOTE: The Available Codec List may be used by a MSC as decision criterion if a codec re-negotiation or modification is attempted. However, this does not preclude that an MSC offers codecs not included in the previous ACL in a codec re-negotiation.

5.7.1.2.7 Encoding of 3GPP OoBTC Indicator

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

3GPP OoBTC Indicator = "a=3gOoBTC"

The 3GPP OoBTC Indicator will be defined in a 3GPP TS and registered at IANA.

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 "Miscellaneous" codec type, shall indicate the Available Codec List (ACL)
- Codecs of "Miscellaneous" codec type, i.e. the telephone-event DTMF codec or the comfort noise codec (RFC3389), may be included as the last codecs in the m-line (indicated by RTP payload type).

5.7.1.2.8 Codec Negotiation Example Sequences

The following figures show examples of codec negotiation for a selection of common call scenarios to highlight the principles agreed in the preceding clause; the sequences are not exhaustive.

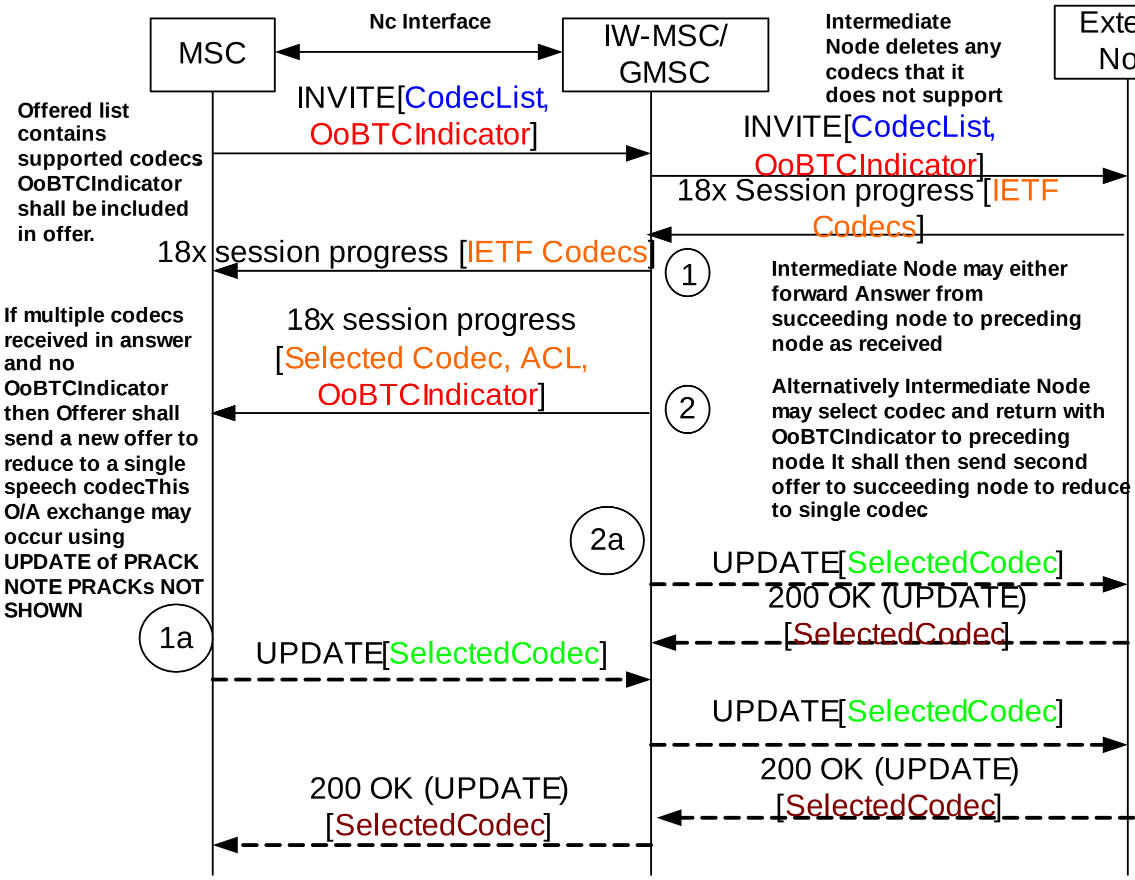

Figure 5.7.1.2.8.1: Mobile Originating Codec Negotiation

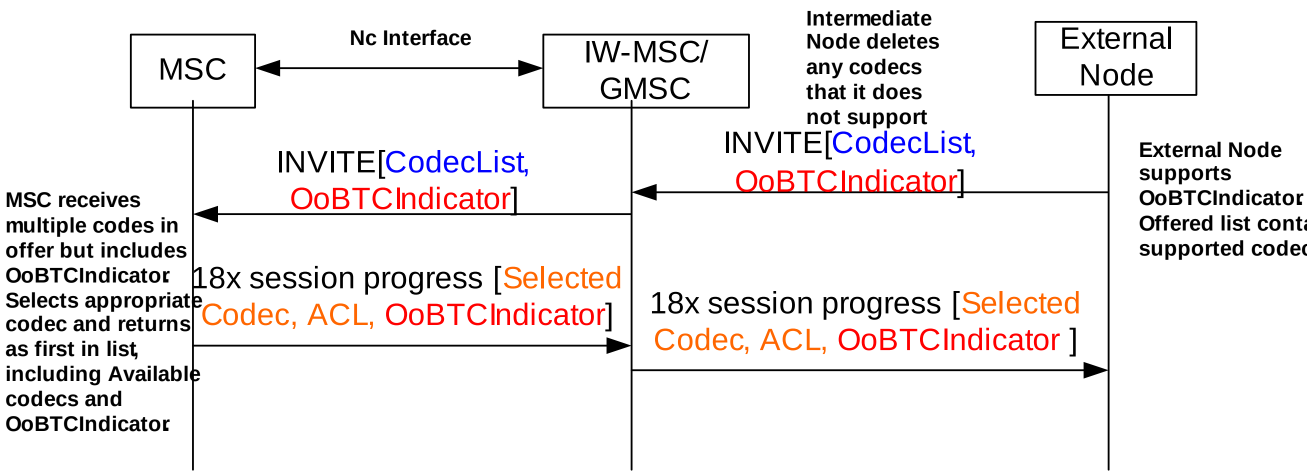

Figure 5.7.1.2.8.2: Mobile Terminating Codec Negotiation – External Node supports OoBTCIndicator

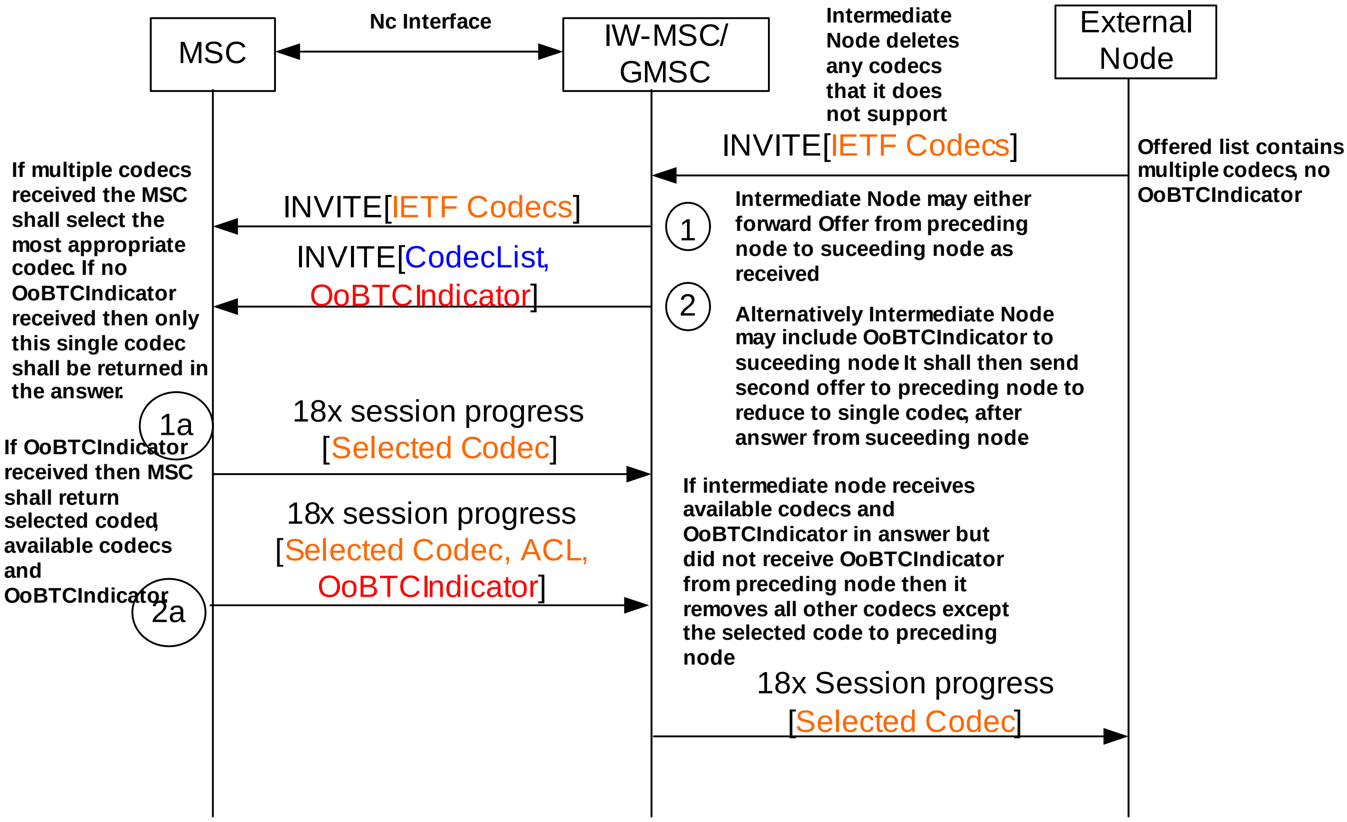

Figure 5.7.1.2.8.3: Mobile Terminating Codec Negotiation – External Node does not support OoBTCIndicator

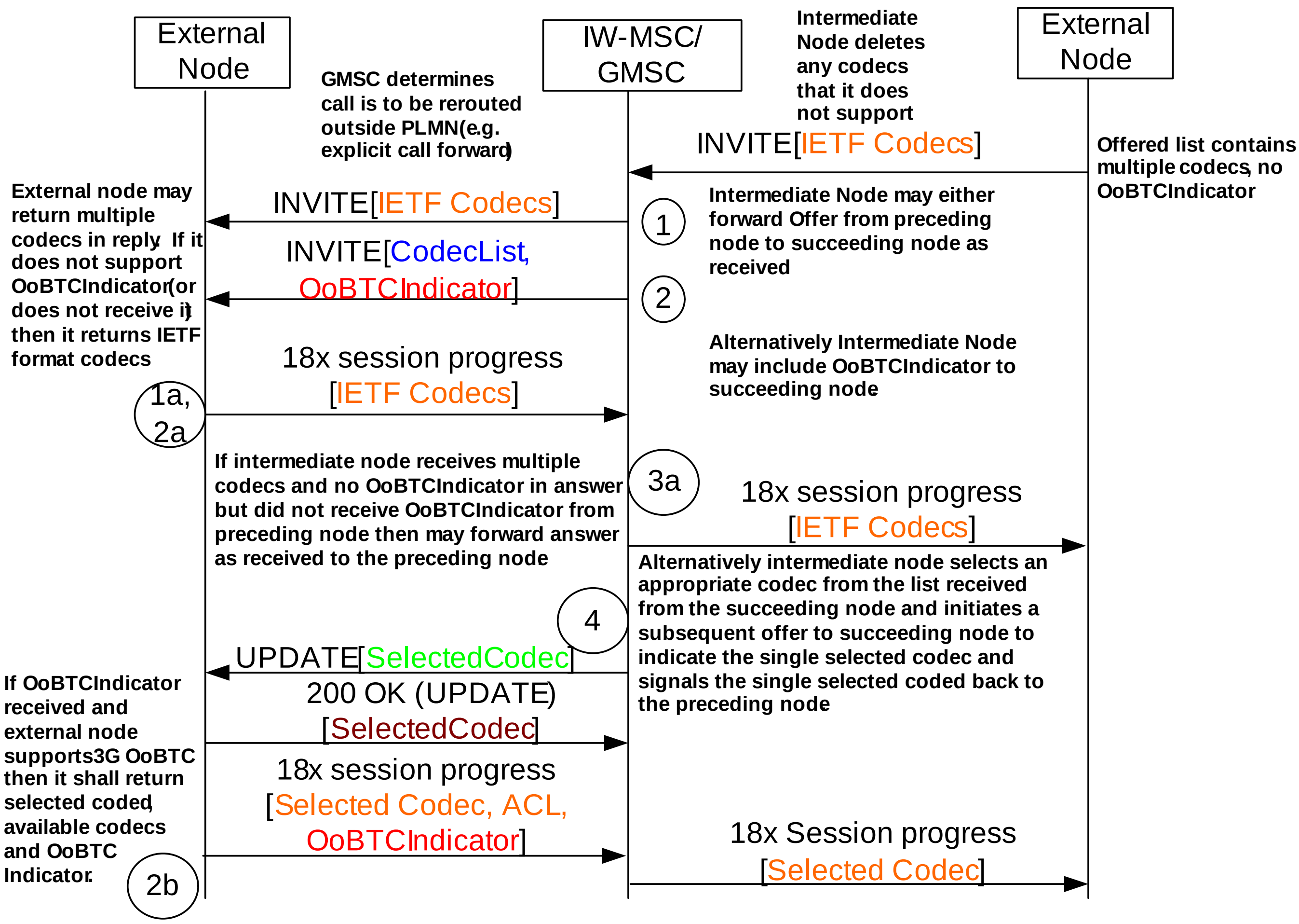

Figure 5.7.1.2.8.4: Codec Negotiation during call forwarding outside of PLMN – external incoming node does not support 3G OoBTCIndicator

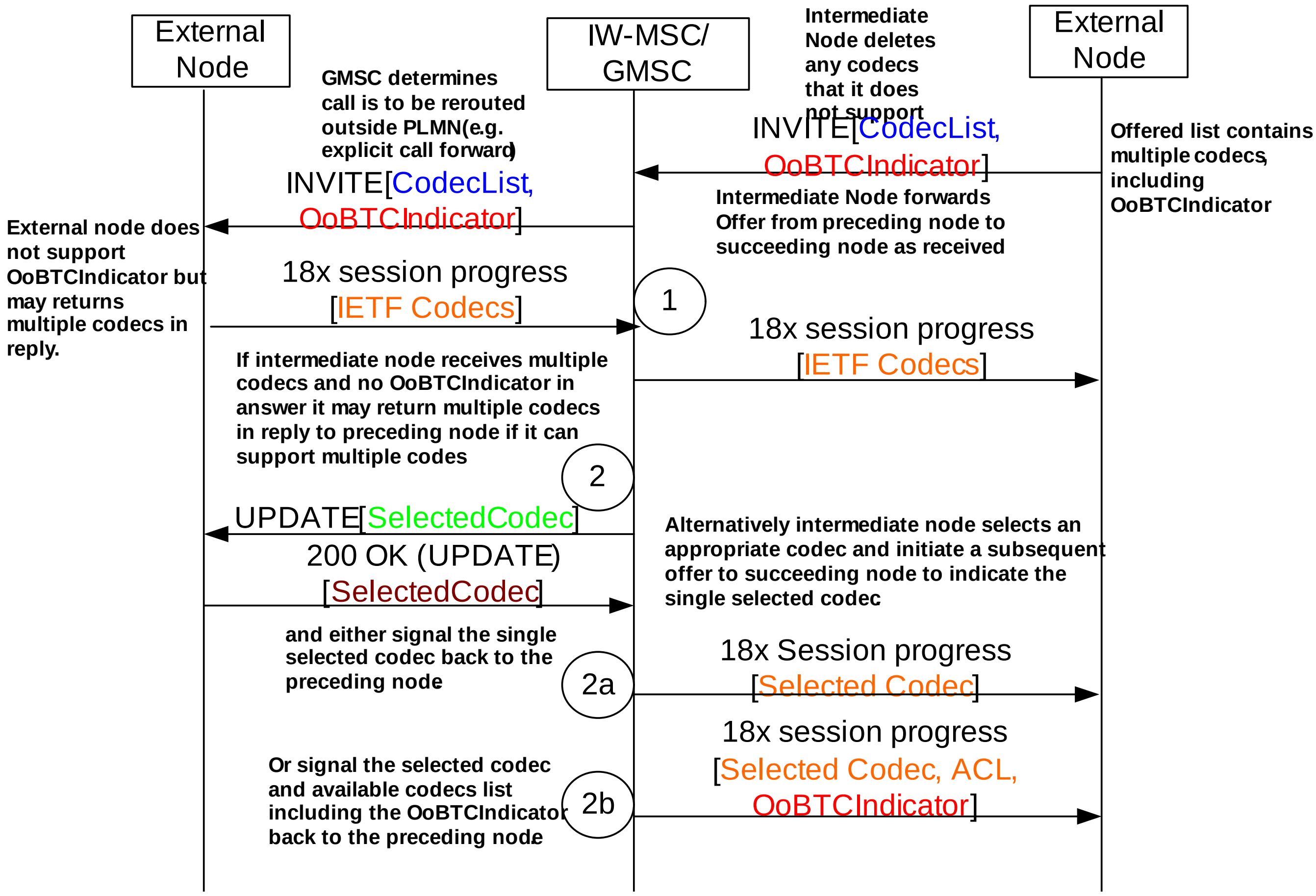

Figure 5.7.1.2.8.5: Codec Negotiation during call forwarding outside of PLMN – external node supports 3G extension

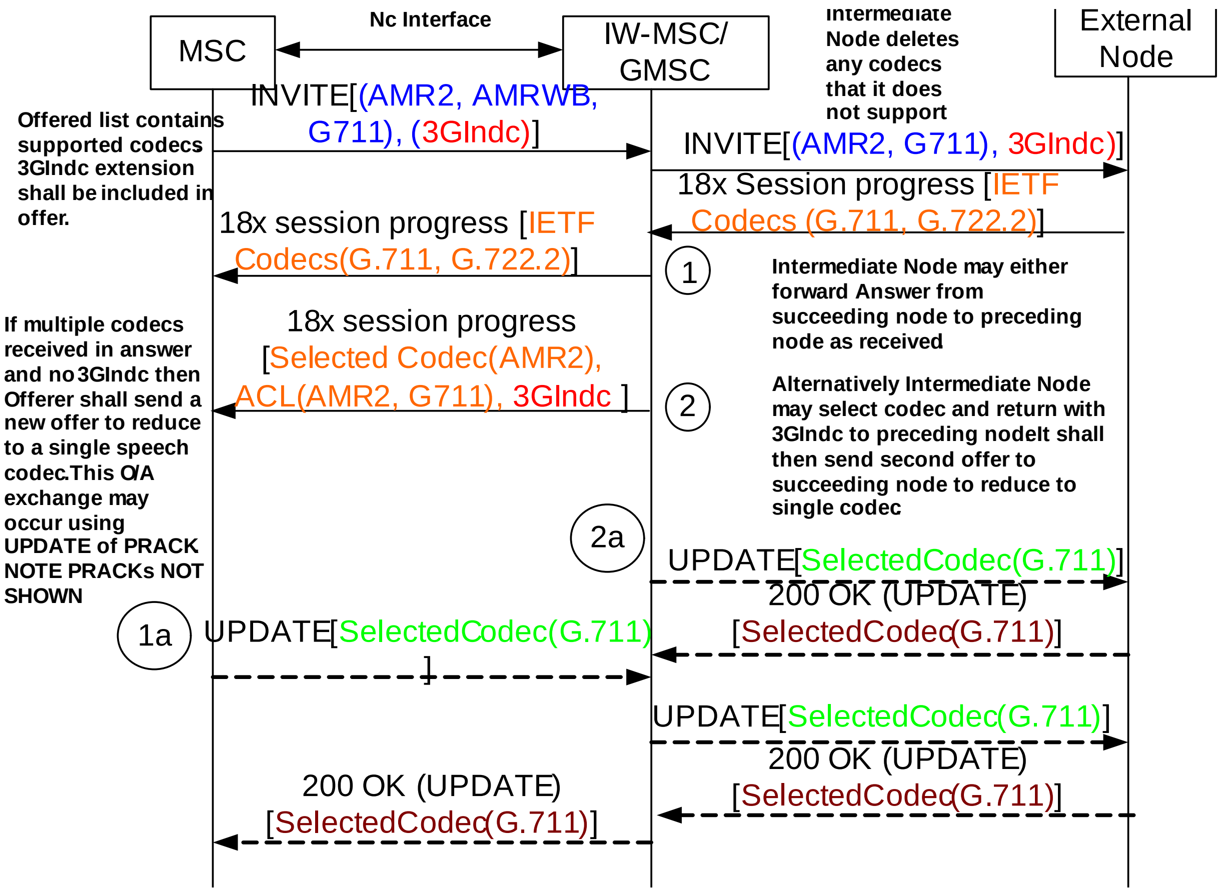

Figure 5.7.1.2.8.6: Mobile Originating Codec Negotiation with specific codec examples

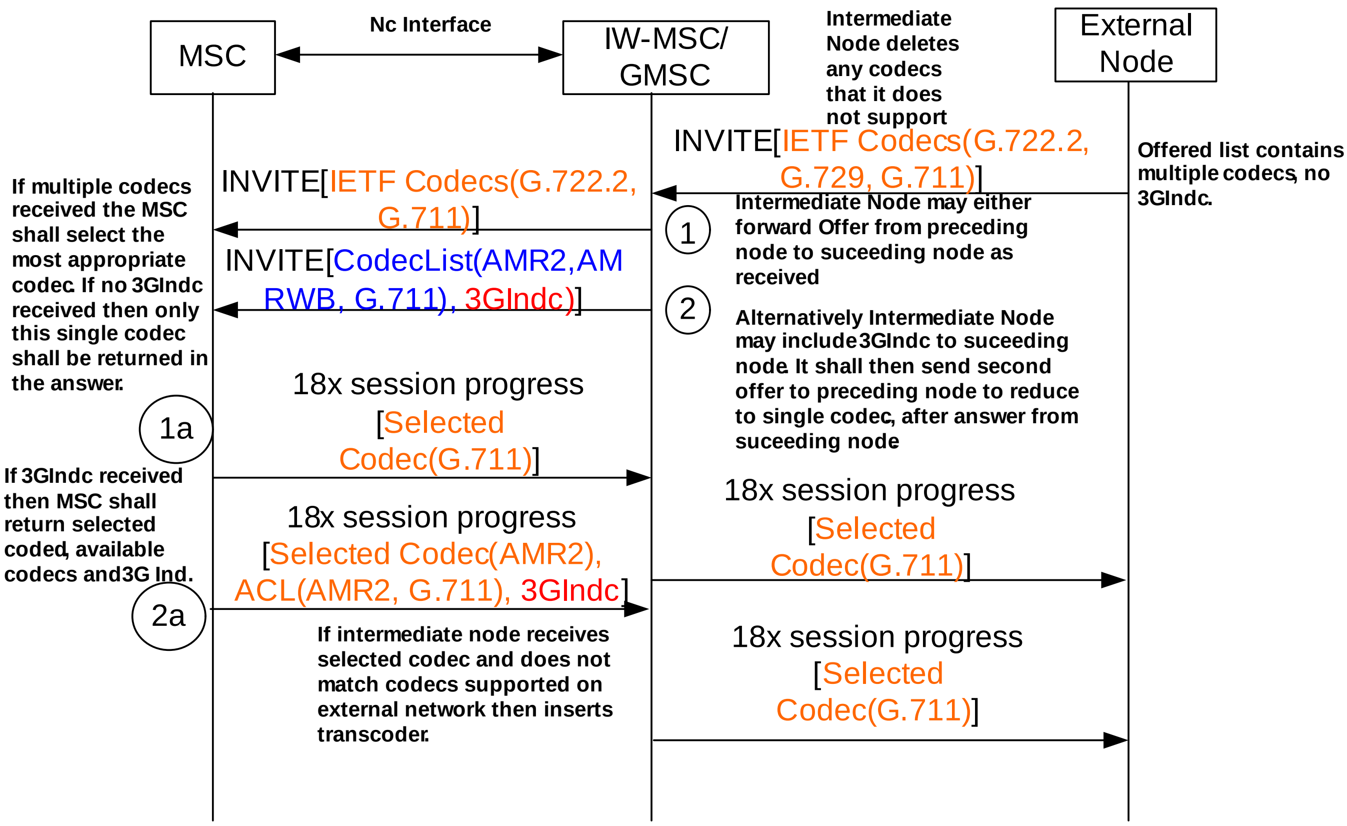

Figure 5.7.1.2.8.7: Mobile Terminating Codec Negotiation with specific codec examples

5.7.2 Offer Answer Rules

5.7.2.1 General

The following are rules that are applied when populating a Session Description Protocol (SDP) media offer or an SDP media answer. A SIP-I signalling endpoint shall initiate an SDP Offer/Answer exchange during call establishment and may initiate an SDP Offer/Answer exchange at any time that the bearer configuration changes, e.g., during handover or invocation of a supplementary service such as 3pty.

Definitions:

- A "Direct Codec" is a codec that can be used without any additional transcoding stage inserted at the MGW subtending the entity providing the SDP. E.g., a direct codec can be AMR or another mobile codec when the end terminal is a mobile station, or G.711 when interworking with the PSTN.
- An "Indirect Codec" is a codec that requires transcoding at the MGW subtending the entity providing the SDP.

5.7.2.2 Rules for Constructing an Offer

The Codec List in the Offer shall contain codecs defined as follows:

- "Direct" codec types that can be used between bearer endpoints without any additional transcoding stage;
- "Indirect" codec types that can be used between bearer endpoints with an additional transcoding stage; and
- "Miscellaneous" codec types unrelated to the primary codec selection process might include, e.g., DTMF and comfort noise.

The Offered Codec List shall contain two sub-lists ordered as: zero or more "direct" codecs plus zero or more "indirect" codecs. A list of zero or more "miscellaneous" codec types, e.g., telephone-events (e.g., DTMF) and CN (comfort noise), which are not used in the process of selecting the primary codec, may follow direct/the indirect codec types.

The direct and indirect codec sub-lists shall be ordered in decreasing preference.

When present, the indirect codec sub-list shall always start with G.711, if G.711 is not a direct codec. However, an entry for G.711 shall appear at most once in the offered codec list.

The offer may contain a list of several direct and indirect codec types.

NOTE: These rules for constructing an SDP Offer enable TrFO in the network by assuring that the network configures the minimum number of transcoders for each session. These rules are needed to enable TrFO in the network and are consistent with IETF RFC 3264 [27], but are not part of IETF RFC 3264 [27]. Other SIP endpoints may not follow the same conventions for prioritizing codecs. As an exception to the aforementioned rules, the offerer could choose to construct an Offered Codec List in a different order from the one described in the above rules, but this is not recommended as the answerer may select a codec that does not minimize the number of transcoders for the session and does not enable TrFO.

5.7.2.3 Rules for Constructing an Answer

The answering signalling endpoint shall, before processing the Offer and before populating the Answer, structure the available codec types on its access into "direct," "indirect", and "miscellaneous" codec types.

The answering signalling endpoint shall then take both structured Codec Lists, the one received in the Offer and the one created locally, into account and shall select the "optimal" codec type for the Answer, which shall be the first codec type in the Answer. The Answer must contain at least one direct or indirect codec from among those listed in the Offer. The endpoints should begin sending voice media using the most preferred media format in the Answer, and may later send using other media formats in the Answer when necessary.

The criteria for the "optimal" codec may depend on operator choices and preferences (local policy), such as Speech Quality, Bit Rate on transport or DSP load (for transcoding) or other.

If the Answer to a subsequent Offer comprises all or a subset of the direct and indirect codecs in the preceding Answer within the dialog, then the IP address, and port information in the SDP Answer should remain the same.

Ideally the Optimal Codec is a direct codec type on both accesses, which results in no transcoding being necessary.

5.7.3 Supported Codecs and Call Events

The following list references the specifications that define the RTP framing procedures that have to be standardized on the Nb interface for the listed codec when SIP-I is the signalling protocol on the Nc interface.

Table 5.7.3.1 Supported payload types

Payload Type Name	References	Applicable Codecs	Support
audio/AMR	draft-ietf-avt-rtp-amr-bis-06.txt [42] 3GPP TS 29.163 Annex B [25]	all AMR codecs in 3GPP TS 26.103 [43]	Mandatory Not all configurations are mandatory. Some configurations are preferred, see 3GPP TS 26.103 [43].
audio/AMR-WB	draft-ietf-avt-rtp-amr-bis-06.txt [42] 3GPP TS 29.163 Annex B [25]	all AMR-WB codecs in 3GPP TS 26.103 [43]	Mandatory if WB is supported. Not all configurations are mandatory, see 3GPP TS 26.103 [43].
audio/GSM-EFR	RFC 3551 [28]	GSM EFR	Optional
audio/PCMA	RFC 3551 [28]	G.711	Mandatory
audio/PCMU	RFC 3551 [28]	G.711	Mandatory
audio/CLEARMODE	RFC 4040 [34]	clear channel data and MUME/SCUDIF	Mandatory
audio/telephone-event	4733 [35]	DTMF	Mandatory
audio/CN	RFC 3389 [41]	comfort noise for CODECs that do not support as part of the CODEC itself such as G.711	Optional

NOTE: Support for MUME may require additions to IETF RFC 4040 [34] since there is currently no mechanism to signal MUME in SDP.

NOTE1: GSM FR and GSM HR (RFC 3551 [28]) codecs will not be supported on SIP-I based Nc, as for BICC based Nc.

Additional codecs or call events may be supported by 3GPP SIP-I nodes on SIP-I based Nc or at the edge of the CS network e.g. to enable interoperation with non-3GPP SIP-I networks, but this is out of the scope of the TR. Examples of such codecs or call events are listed below.

Table 5.7.3.2 Examples of additional payload types

Payload Type Name	References	Applicable Codecs
image/T38	ITU-T Rec. T.38	G3 facsimile
audio/T38	ITU-T Rec. T.38	G3 facsimile
text/T140	RFC 4103 [63] ITU-T Rec. T.140	GTT
text/RED	RFC 4102 RFC 2198	Redundant GTT
audio/G723	RFC 3551	MUME audio option G.723.1
audio/G729 audio/G729D audio/G729E	RFC 3551	NGN codecs
video/H261	RFC 3551	MUME video option H.261
video/H263	RFC 3551	MUME video option H.263
video/mp4v-es	RFC 3640 [62]	MUME video option mpeg4
Video/H264	RFC 3984 [69]	MUME video option mpeg4

5.8 Usage of URI schemas in the Nc interface

Several kinds of uniform resources identifiers can be used in the SIP protocol, e.g. SIP and SIPS URI defined in IETF RFC 3261 [30], tel URI defined in IETF RFC 3966 [31], etc.

Since the user and service are always identified by an E.164 number in the bearer independent CS network, tel-URI is more suitable because it is always hard to say which domain is responsible for it and how to fill in the host part of the SIP or SIPS URI. The same thing applies for the calling party number, the called party number, etc.. For these fields, tel-URI shall be supported.

Therefore, receipt of tel-URI and SIP-URI (user=phone) shall be supported, sending of tel-URI shall be supported, and sending of SIP-URI (user=phone) may be supported.

6 IP Version

IPv4, IPv6 or both IPv4 and IPv6 could be supported in the Nc, Mc and Nb interface.

As IPv4 is widely supported in the current IP based network for years, and almost all of the currently in-service MSC servers are based on IPv4, Therefore IPv4 shall be supported in the Nc, Mc and Nb interface.

IPv6 is the basic IP version in the IMS network. So IPv6 may be supported in the Nc, Mc and Nb interface as well.

7 Security

The supported security mechanisms for IP signalling transport for a SIP-I based Nc interface are described in 3GPP TS 33.210 [32].

8 Call Establishment

NOTE: All message sequence charts in this clause are examples. All valid call establishment message sequences can be derived from the example message sequences and associated message pre-conditions.

8.1 Basic Mobile Originating Call

8.1.1 Basic Mobile Originating Call Establishment

The mobile originating call shall be established in accordance with 3GPP TS 23.108 [39]. The following clauses describe the additional requirements for the CS core network. The Offer/Answer Model with the Session Description Protocol (SDP) and Codec Negotiation principle as defined in Clause 5.7 shall be applied for SDP exchange and media negotiation with IETF RFC 3264 [39].

8.1.1.1 MGW selection

The MSC server shall select an MGW for the bearer connection before it performs the access bearer assignment or the network side connection point reservation. This shall happen before sending the INVITE message.

8.1.1.2 Initial INVITE message

The MSC server shall send the initial INVITE message after getting the bearer characteristics and the bearer address, before or after the access bearer assignment is completed. The MSC server provides the bearer characteristics and the bearer address to the succeeding node in the SDP of the initial INVITE message. The initial INVITE message shall encapsulate the IAM message. If the access bearer assignment has not been completed, the MSC server shall indicate that the local precondition has not been met.

8.1.1.3 Network side bearer establishment

The MSC server shall either select bearer characteristics or request the MGW to select and provide the bearer characteristics for the network side bearer connection before sending the INVITE message. The MSC server shall use the Reserve RTP Connection Point procedure (bullet 1 in figure 8.1.1.11.2). Within this procedure, the MSC server shall indicate the local codec(s) and request a local IP address and UDP port from the MGW. The local IP address and UDP port are used by the MGW to receive user plane data from the succeeding MGW.

The MGW shall reply to the MSC server with the selected local codec(s) and the selected local IP address and UDP port.

The MSC server shall send this information in the INVITE (bullet 2 in figure 8.1.1.11.2) to the succeeding node.

After the succeeding node has provided the SDP answer, the MSC server uses the Configure RTP Resources to request the MGW to configure the bearer (bullet 3 in figure 8.1.1.11.2).

8.1.1.4 Access bearer assignment

The access bearer assignment is defined in the clause 6.1.1.4 of 3GPP TS 23.205 [2].

8.1.1.5 Through-Connection

During any one of the Prepare Bearer and Reserve Circuit procedures, the MSC server will use the Configure RTP Connection Point procedure to request the MGW to configure the bearer terminations so that the bearer will be bidirectional through-connected (bullet 3 and bullet 4 or 5 in figure 8.1.1.11.2).

When the MSC server receives the answer indication, it requests the MGW to both-way through-connect the bearer using the Change Through-Connection procedure (bullet 8 in figure 8.1.1.11.2).

8.1.1.6 Confirmation of bearer establishment

If the initial INVITE message which was sent to the succeeding node indicated that the local precondition has not been met, the MSC server sends the UPDATE message with local precondition is met when the access bearer assignment has been completed (bullet 6 in figure 8.1.1.11.2).

8.1.1.7 Interworking function

The Interworking function is FFS.

8.1.1.8 Codec handling

The MGW may include a speech transcoder based upon the speech coding information provided to each bearer termination.

8.1.1.9 Voice Processing function

The Voice Processing function is FFS.

8.1.1.10 Failure handling in MSC server

The Failure handling in MSC server is FFS.

8.1.1.11 Example

Figure 8.1.1.11.1 shows the network model for the mobile originating call. The "squared" line represents the call control signalling. The "dotted" line represents the bearer control signalling (not applicable in A/Gb mode for the A-interface) and the bearer. The MSC server seizes one context with two terminations in the MGW. The bearer termination T1 is used for the bearer towards the RNC/BSC and the bearer termination T2 is used for the bearer towards the succeeding MGW.

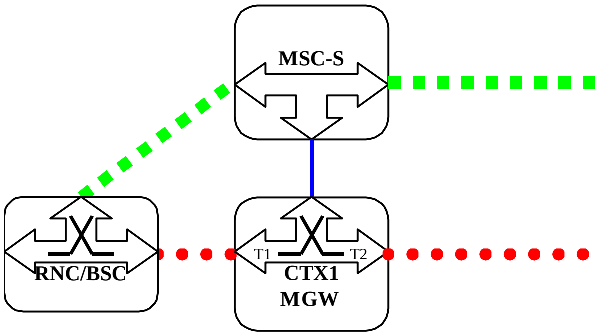

Figure 8.1.1.11.1: Basic Mobile Originating Call (network model)

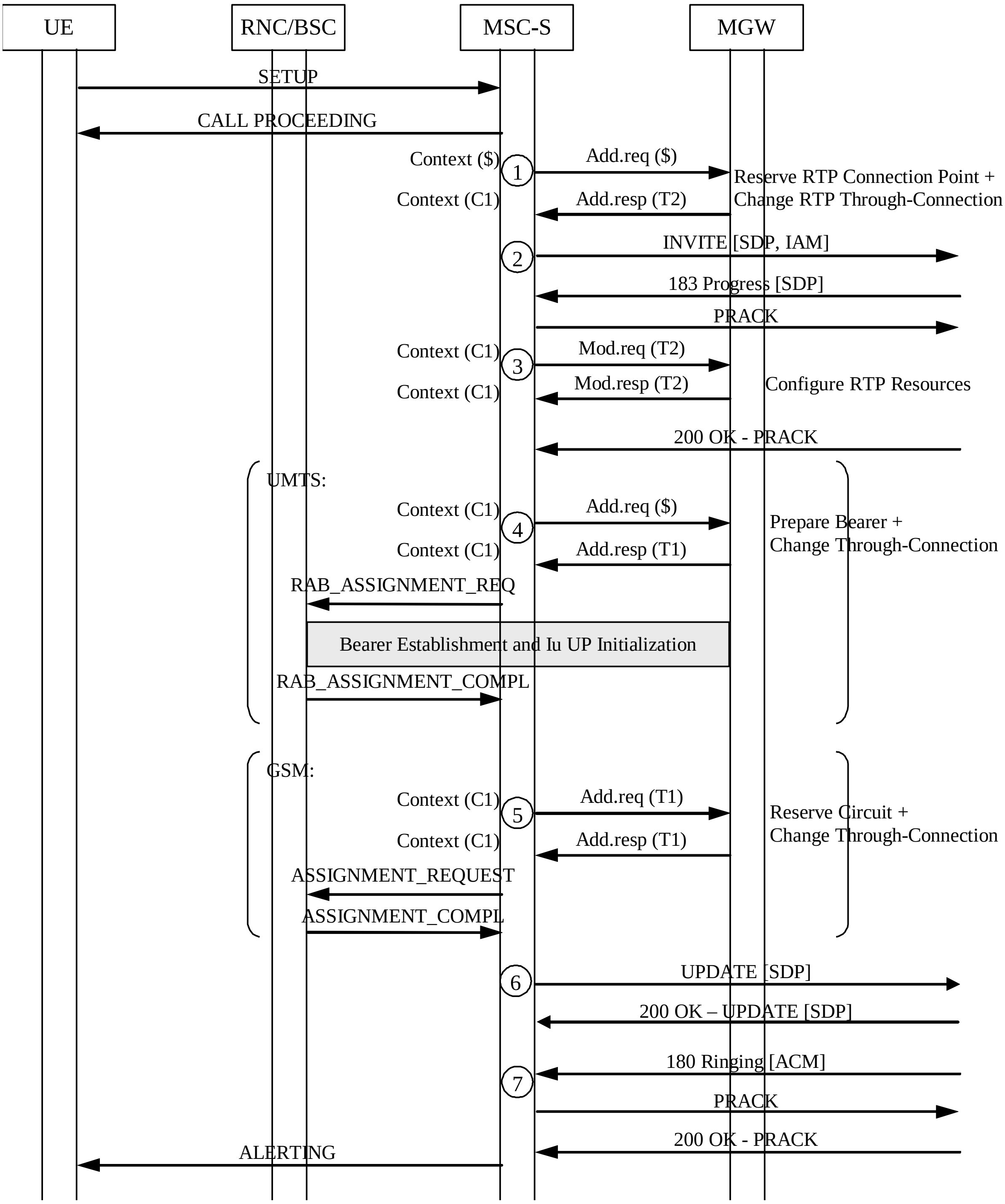

Figure 8.1.1.11.2/1: Basic Mobile Originating Call with early originating access bearer assignment (message sequence chart).

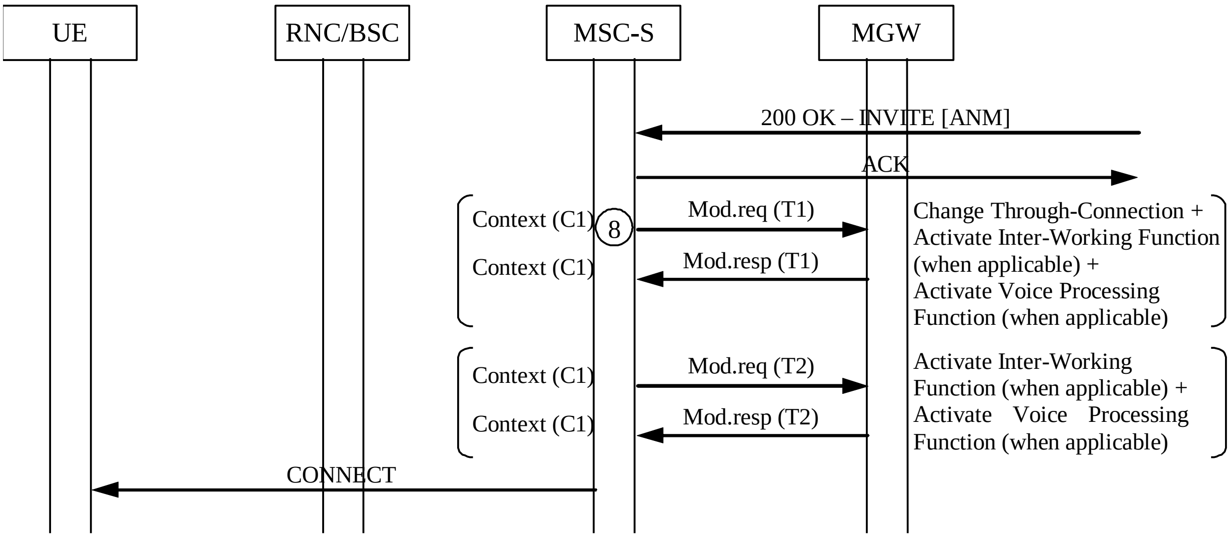

Figure 8.1.1.11.2/2: Basic Mobile Originating Call with early originating access bearer assignment (message sequence chart).

8.1.2 Originating Call Establishment For Iu Interface on IP

If IuCS on IP is supported by the MSC server, the Core Network side procedures described in 8.1.1 shall apply. For the access side termination, the exchange of IP addresses via call control procedures defined in the clause 6.1.3 of 3GPP TS 23.205 [2] shall apply.

If the bearer transport is IP and IuUP mode is Support, the MGW shall use the source IP address and UDP Port of the IuUP Init packet received from the radio access network as the destination address for subsequent downlink packets.

The sequence is shown in Figure 8.1.2.1.

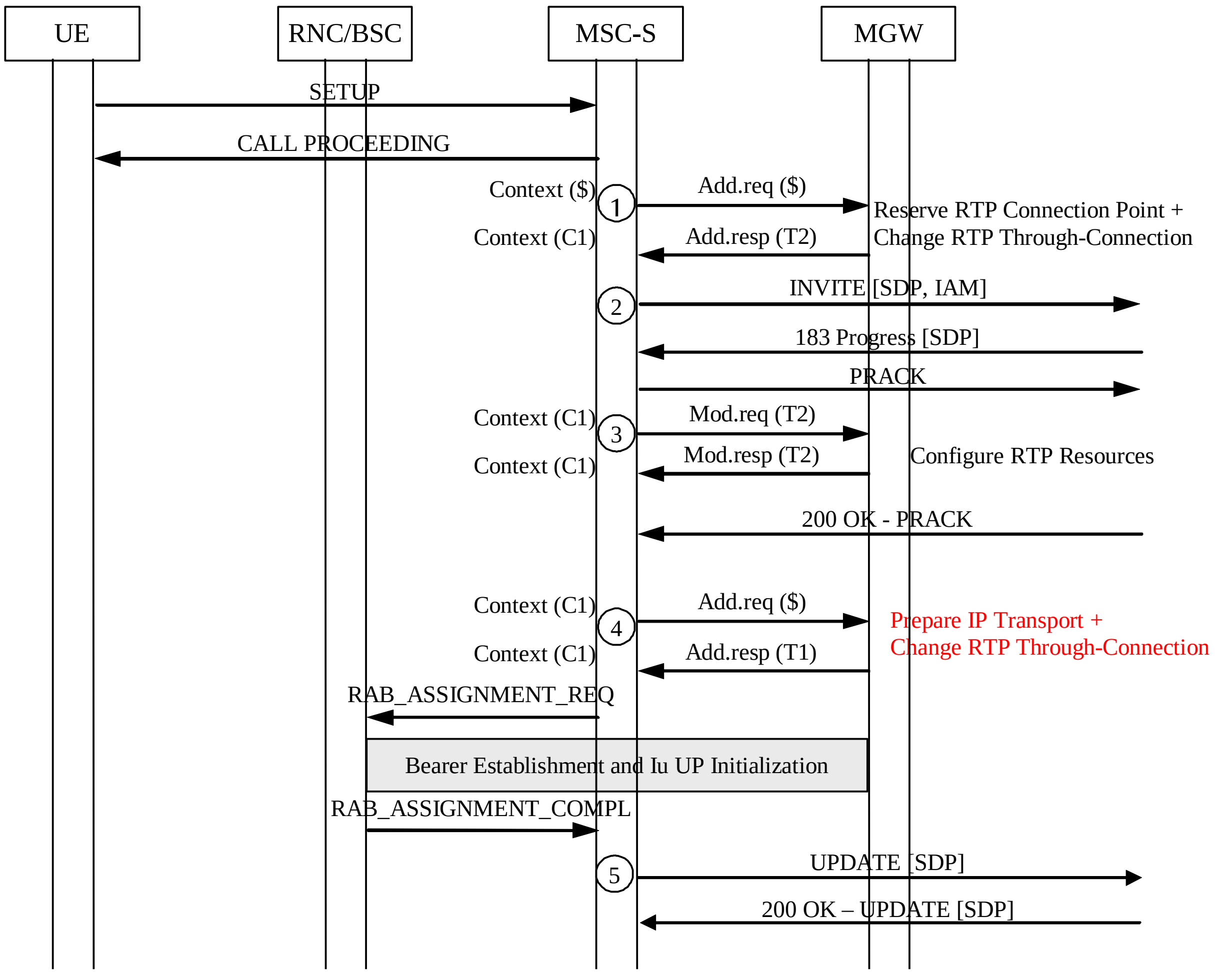

Figure 8.1.2.1: Call Establishment for Iu on IP

8.2 Basic Mobile Terminating Call

8.2.1 Basic Mobile Terminating Call Establishment

8.2.1.1 GMSC server

NOTE: Mobile terminating calls without MGW allocated by the GMSC is FFS.

8.2.1.1.1 MGW selection

The GMSC server shall select an MGW for the bearer connection before it performs the incoming side bearer establishment or the outgoing side bearer establishment. This shall happen before sending the initial INVITE message.

8.2.1.1.2 Initial INVITE message

The GMSC shall provide the bearer characteristics and the bearer address for the outgoing side bearer connection to the succeeding node in the SDP of the INVITE message.

8.2.1.1.3 Outgoing side bearer establishment

The GMSC server shall select bearer characteristics for the outgoing side bearer connection before it performs the Reserve RTP Connection Point procedure. After the succeeding node has provided a bearer address in SDP of a response message the MSC server uses the Configure RTP Resources procedure to request the MGW to configure the bearer towards the destination MGW. The MSC server provides the MGW with the bearer address and the bearer characteristics.

8.2.1.1.4 Incoming side bearer establishment

The GMSC server shall request the MGW to prepare for the incoming side bearer establishment using the Reserve RTP Connection Point procedure. The GMSC server requests the MGW to provide a bearer address and the bearer characteristics. The GMSC server also provides the MGW with the bearer characteristics that was received from the preceding node in the INVITE message. After the MGW has replied with the bearer address and the bearer characteristics, the GMSC server shall send a SDP answer with the bearer address and the bearer characteristics to the preceding node.

NOTE: The incoming side bearer establishment may take place either before or after HLR interrogation.

8.2.1.1.5 Through-Connection

During the Reserve RTP Connection Point and Configure RTP Resources procedures, the GMSC server will use the Change RTP Through-Connection procedure to request the MGW to both-way through-connect the bearer termination the GMSC server selects an MGW for the bearer connection.

8.2.1.1.6 Confirmation of bearer establishment

Handling of precondition by the GMSC FFS.

8.2.1.1.7 Voice Processing function

The Voice Processing function is FFS.

8.2.1.1.8 Failure handling in GMSC server

The Failure handling in MSC server is FFS.

8.2.1.2 MSC server

8.2.1.2.1 Paging

The Paging is defined in the clause 6.1.1.4 of 3GPP TS 23.205 [2].

8.2.1.2.2 Call setup

The MSC server indicates to the UE in the SETUP message that early access bearer assignment is used in order to establish the bearer end-to-end before the UE starts alerting. The MSC server indicates to the UE in SETUP message that early access bearer assignment is used if the INVITE message indicates the remote pre-condition has not been met or a notification of successful bearer establishment in the network side has not been received from the MGW before sending the SETUP message (bullet 3 in figure 8.2.1.2.14.2/1).

8.2.1.2.3 MGW selection

The MSC server shall select an MGW for the bearer connection before it performs the network side bearer establishment or the access bearer assignment. This happens at latest after the UE has sent the Call Confirmed message. For GSM, if performing Service based handover (see 3GPP TS 48.008 [64]) the MSC Server may omit MGW selection at this time.

8.2.1.2.4 Network side bearer establishment

The MSC server requests the MGW to prepare for the network side bearer establishment using the Reserve RTP Connection Point and Configure RTP Resources procedure. The MSC server requests the MGW to provide a bearer address and to notify when the bearer is established (bullet 4 in figure 8.2.1.2.14.2/1). The MSC server also provides the MGW with the bearer characteristics that was received from the preceding node in the INVITE. After the MGW has replied with the bearer address, the MSC server also sends a 183 Progress response with the bearer address and the binding reference to the preceding node.

8.2.1.2.5 Access bearer assignment

The access bearer assignment may be started when the remote pre-condition has been met and the notification of successful bearer establishment in the network side has been received from the MGW.

The access bearer assignment is defined in the clause 6.2.1.2.5 of 3GPP TS 23.205 [2].

8.2.1.2.6 Framing protocol initialisation

There is no specific framing protocol initialisation requirement in SIP-I based CS network.

8.2.1.2.7 Called party alerting

For a speech call, when the MSC server receives an Alerting message, it requests the MGW to provide a ringing tone to the calling party using the Send RTP Tone procedure (bullet 10 in figure 8.2.1.2.14.2/1).

NOTE: Other kind of tones may be provided to the calling party at an earlier stage of the call establishment.

8.2.1.2.8 Called party answer

For a speech call, when the MSC server receives a Connect message, it requests the MGW to stop providing the ringing tone to the calling party using the Stop RTP Tone procedure (bullet 12 in figure 8.2.1.2.14.2/2).

8.2.1.2.9 Through-Connection

During the Reserve RTP Connection Point, Prepare Bearer and Reserve Circuit procedures, the MSC server will use the Change RTP Through-Connection procedure to request the MGW to through-connect the bearer terminations so that the bearer will be not through connected (bullet 4, and bullet 8 or 9 in figure 8.2.1.2.14.2).

When the MSC server receives the Connect message, it requests the MGW to both-way through-connect the bearer using the Change RTP Through-Connection procedure (bullet 12 in figure 8.2.1.2.14.2/2).

8.2.1.2.10 Interworking function

The Interworking Function is FFS.

8.2.1.2.11 Codec handling

Codec handling in MSC server is FFS.

8.2.1.2.12 Voice Processing function

The Voice Processing Function is FFS.

8.2.1.2.13 Failure handling in MSC server

The Failure handling in MSC server is FFS.

8.2.1.2.14 Example - the GMSC selects a MGW

Figure 8.2.1.2.14.1 shows the network model for the basic mobile terminating call. The "squared" line represents the call control signalling. The "dotted" line represents the bearer control signalling (not applicable in A/Gb mode for the A-interface) and the bearer. The MSC server seizes one context with two bearer terminations in MGWb. The bearer termination T1 is used for the bearer towards the RNC/BSC and the bearer termination T2 is used for the bearer towards the GMSC server selected MGWa. The GMSC server seizes one context with two bearer terminations in MGWa. The bearer termination T3 is used for the bearer towards the MSC server selected MGWb and the bearer termination T4 is used for the bearer towards the preceding MGW.

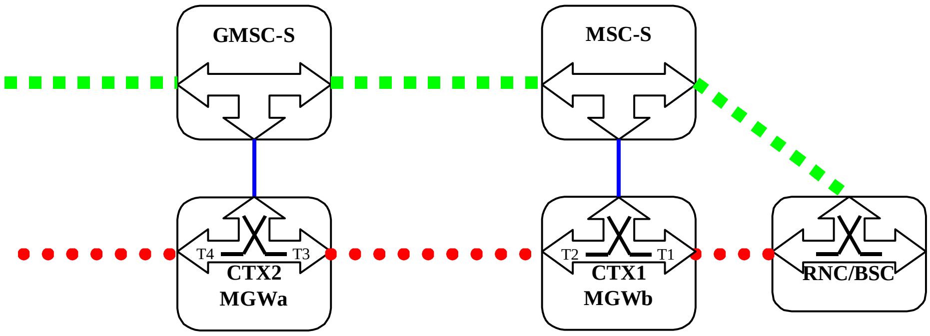

Figure 8.2.1.2.14.1: Basic Mobile Terminating Call Forward Bearer Establishment (network model)

Figure 8.2.1.2.14.2 shows the message sequence example for the basic mobile terminating call. In the example the GMSC server requests seizure of the outgoing side bearer termination and establishment of the bearer when the Bearer Information message is received from the MSC server. After the outgoing side bearer termination is seized the GMSC server requests seizure of the incoming side bearer termination. The MGW sends a notification of an established incoming side bearer. The MSC server requests seizure of the network side bearer termination when Call Confirmed message is received from the UE. The MGW sends a notification of an established network side bearer. The MSC server requests seizure of the access side bearer termination. For a speech call the MSC server requests MGW to provide a ringing tone to the calling party at alerting. At answer the MSC server requests MGW to both-way through-connect the bearer. For a speech call the MSC server requests MGW to stop the ringing tone to the calling party at answer. When the MSC server receives an answer indication, it shall request the possible activation of the interworking function in both bearer terminations. The (G)MSC server shall request the possible activation of the voice processing functions for the bearer terminations.

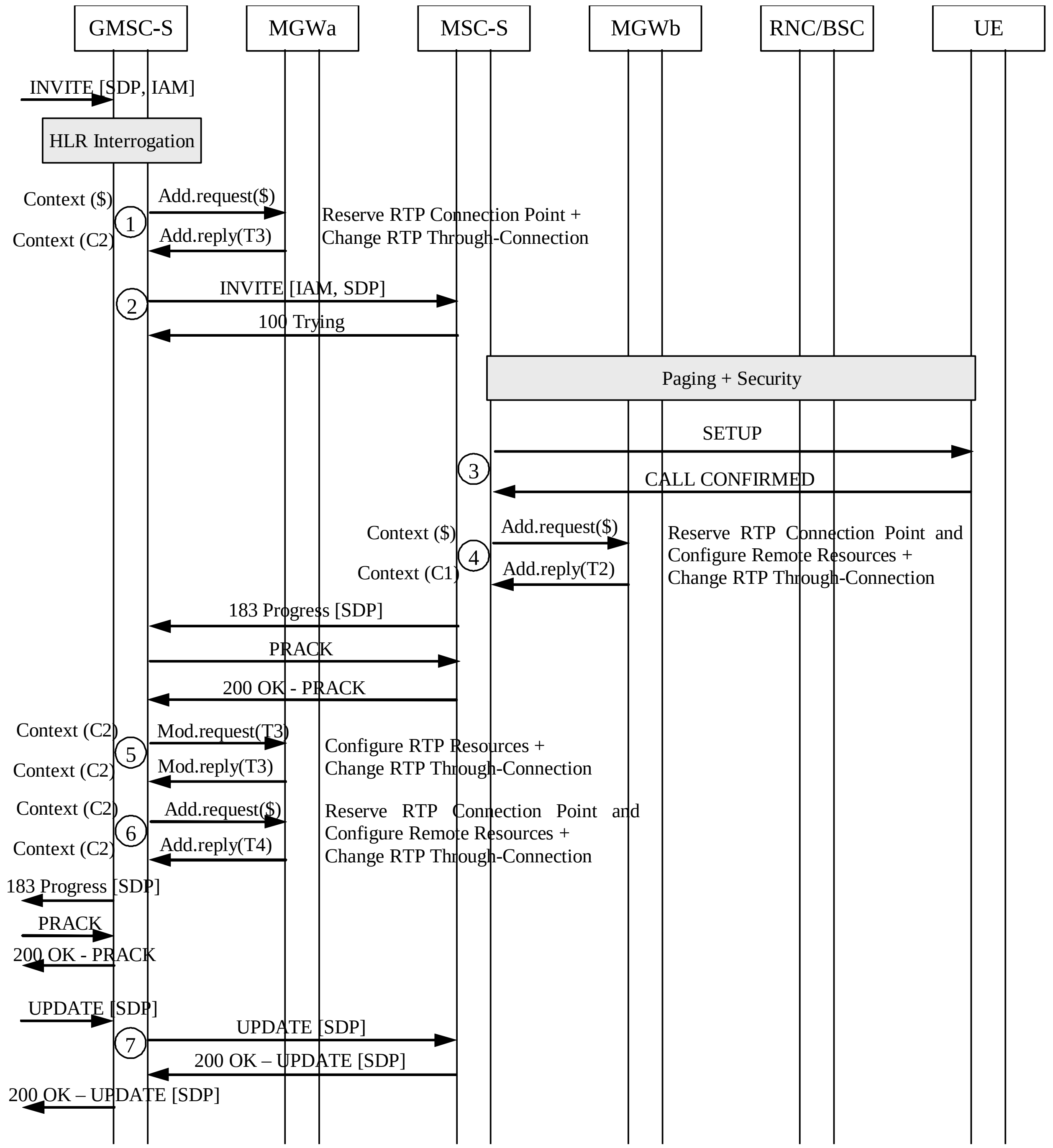

Figure 8.2.1.2.14.2/1: Basic Mobile Terminating Call, GMSC selects a MGW (message sequence chart)

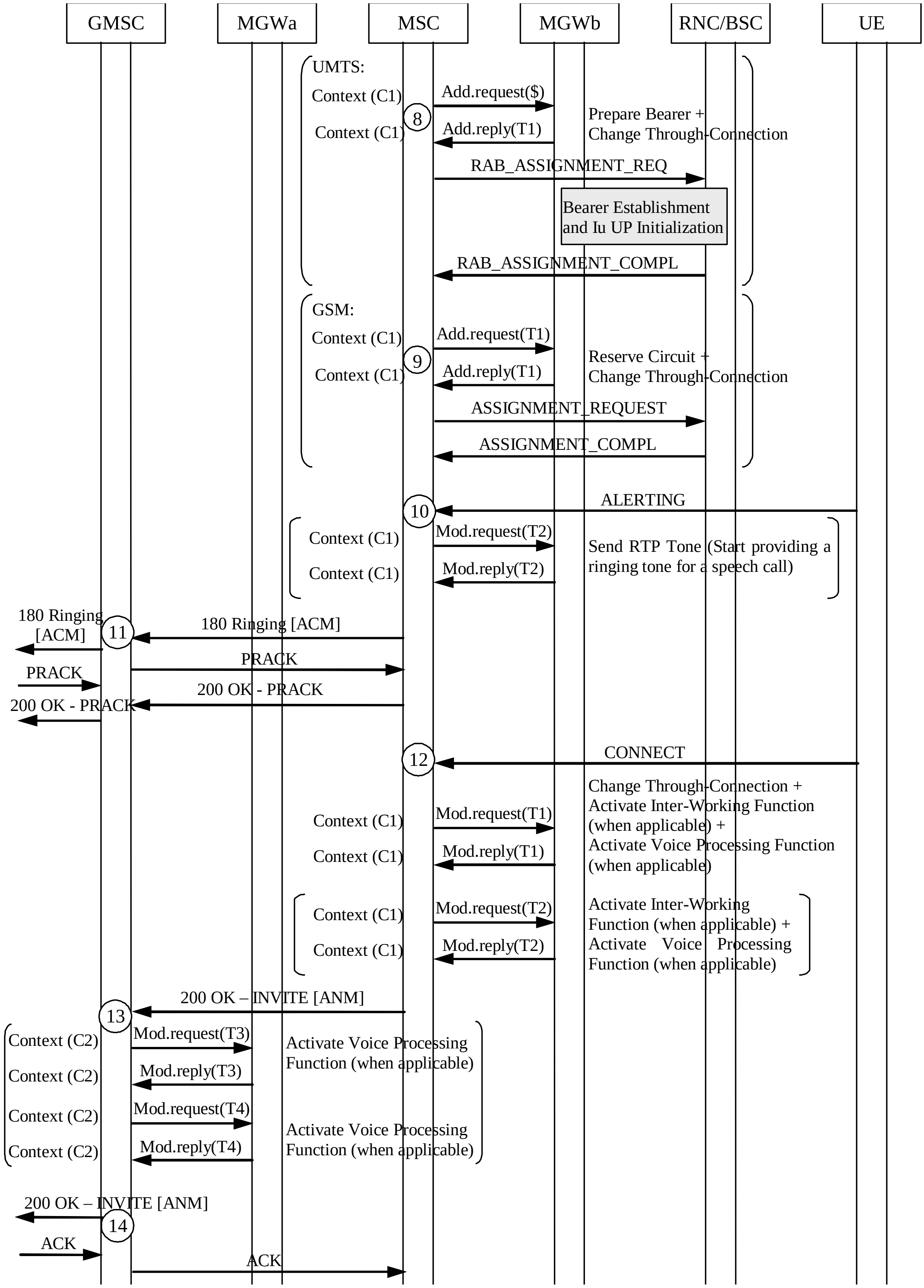

Figure 8.2.1.2.14.2/2: Basic Mobile Terminating Call, GMSC selects a MGW (message sequence chart continue)

8.2.2 Terminating Call Establishment For Iu Interface on IP

If IuCS on IP is supported by the MSC server, the Core Network side procedures described in Clause 8.2.1 shall apply. For the access side termination, the exchange of IP addresses via call control procedures defined in the Clause 6.2.3 of 3GPP TS 23.205 [2] shall apply.

If the bearer transport is IP and IuUP mode is Support, the MGW shall use the source IP address and UDP Port of the IuUP Init packet received from the radio access network as the destination address for subsequent downlink packets.

The sequence is shown in Figure 8.2.2.1.

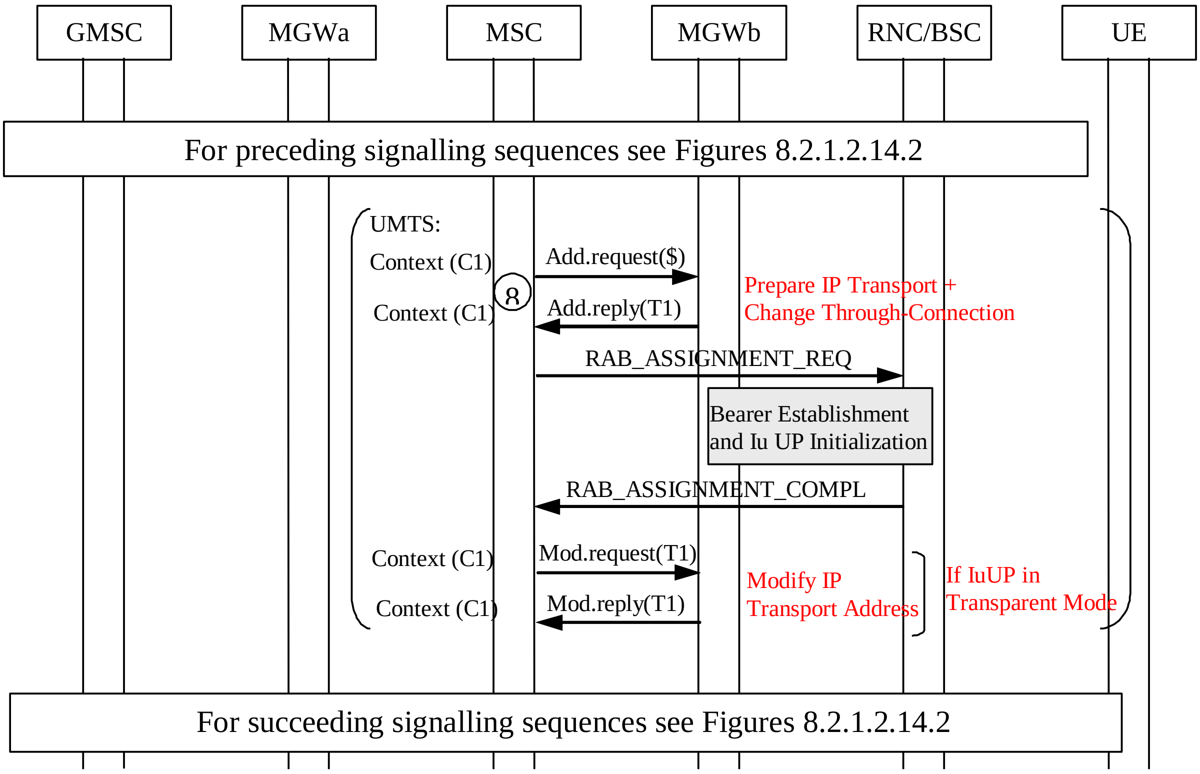

Figure 8.2.2.1: Call Establishment for Iu on IP

9 Mid Call Codec Negotiation

The section is to define the mid call codec negotiation flow examples.

9.1 Network Initiated

The section is to define the network initiated mid call codec negotiation flow examples.

9.2 User Initiated

The section is to define the user initiated mid call codec negotiation flow examples.

10 Call Clearing

The terms "incoming" and "outgoing" in the following text refers to the direction of propagation of the release indication, not to the direction of original call establishment.

During call establishment, the call can be released by sending a BYE or CANCEL request in the direction of the original call establishment, or by a failure response in the opposite direction of the original call establishment. For established call, the call can be released by BYE request in the direction of the original call establishment or in the opposite direction of the original call establishment.

The term "release indication" in the following text refers to any message which is to release the call, i.e. it can be the BYE or CANCEL request, or the failure response to the initial INVITE request.

NOTE: All message sequence charts in this clause are examples. All valid call clearing message sequences can be derived from the example message sequences and associated message pre-conditions.

10.1 Network Initiated

10.1.1 GMSC server

If the following release indication is received from the preceding/succeeding node, a corresponding procedure shall be performed. The following procedures have been proposed:

Alternative 1:

1. If the release indication is a CANCEL request from the preceding node during call establishment, the MSC server shall send a 200 OK response for the CANCEL request and a 487 Request Terminated response for the initial INVITE request to the preceding node, initiate a CANCEL request to the succeeding node, and perform the procedure described below the list of bullets, to release the resources reserved for the call. If the GMSC receives a response for the initial INVITE request instead of a 200 OK response for the CANCEL from the succeeding node, the GMSC server shall send an ACK request to the succeeding node. If the response from the succeeding node is a successful response, GMSC server shall initiate a BYE request to the succeeding node to release the call.

2. If the GMSC receives a BYE request (during early dialog i.e. after sending the provisional 18x response) from the preceding node during call establishment, the GMSC server shall send a 200 OK response for the BYE request and a 487 Request Terminated response for the initial INVITE request to the preceding node, initiate a BYE request to the succeeding node, and perform the procedure described below the list of bullets, to release the resources reserved for the call. If the GMSC receives a response for the initial INVITE request instead of a 200 OK response for the BYE from the succeeding node, the GMSC server shall send an ACK request to the succeeding node and wait for a final response for the BYE request.

   NOTE: these procedures only apply if the GMSC acts as a B2BUA and does not forward immediately the 2xx final response to the initial INVITE.

3. If the GMSC server receives a CANCEL request from the preceding node after having returned a final response to the initial INVITE request, the CANCEL shall be acknowledged but no further action taken.

4. If the release indication is a BYE request from the preceding/succeeding node during confirmed dialog, the MSC server shall send a 200 OK response for the BYE request to the preceding/succeeding node, perform the procedure described below the list of bullets, and forward the BYE request to the succeeding/preceding node.

5. If the message is a failure response for the initial INVITE request for an establishing call from the succeeding node, the GMSC server shall send an ACK for the failure response to the succeeding node, perform the procedure described below, and then forward the failure response to the preceding node.

In above cases, the GMSC server shall perform a procedure to release seized resource: the GMSC server releases any MGW allocated resources for the incoming side and for the outgoing side. If any resources were seized in the MGW, the GMSC server shall use the Release RTP Termination procedures to indicate to the MGW to remove the incoming side bearer termination and that the bearer can be released towards the preceding MGW.

Alternative 2:

NOTE: This Alternative has been proposed as late alternative and is not fully endorsed by CT4.

1. If the GMSC receives a CANCEL request from the preceding node, the GMSC shall send a 200 OK response for the CANCEL request to the succeeding node. If the GMSC did not yet receive a 2xx response for the initial

INVITE request from the succeeding node, the GMSC shall then initiate a CANCEL request to the succeeding node.

NOTE: The GMSC server will also receive a 487 Request Terminated response to the cancelled INVITE, that will be handled as described in bullet 3, unless the CANCEL is crossing with a final SIP response due to a race condition. In this case, the sender of the CANCEL can use a BYE to terminate the call according to SIP procedures. The MGW resources will be released by the GMSC due to those signalling interactions, as described in the bullets below.

2. If the GMSC server receives a BYE request from the preceding/succeeding node, the GMSC shall perform the procedure described below the list of bullets to release seized MGW resources, and forward the BYE request to the succeeding/preceding node. The MSC server will then receive a 200 OK(BYE) response from the succeeding/preceding node and shall forward this 200 OK(BYE) response to the preceding/succeeding node.

3. If the GMSC server receives a final error response (4xx, 5xx or 6xx) for the initial INVITE request from the succeeding node, the GMSC server shall perform the procedure described below the list of bullets to release seized MGW resources, and then forward the final error response to the preceding node. The MSC server will then receive an ACK from the preceding node and shall forward this ACK to the succeeding node.

When indicated in the above list of bullets, the GMSC server shall perform the following procedure to release any MGW allocated resources for the incoming side and for the outgoing side. If any resources were seized in the MGW, the GMSC server shall use the Release RTP Termination procedures to indicate to the MGW to remove the incoming side bearer termination and that the bearer can be released towards the preceding MGW.

NOTE: The handling of SIP redirect responses, i.e. SIP responses of "3xx" type, needs to be added both to alternative 1 and alternative 2. Possible GMSC behaviour that may be further investigated includes: The GMSC forwards those responses, or the GMSC follows the redirects, or the GMSC terminates the call. The appropriate handling may depend on operator's policy and trust relationships to the network from where the 3xx response is received.

10.1.2 MSC server

The network initiated call clearing shall be performed in accordance with 3GPP TS 23.108 [18]. The following clauses describe the additional requirements for the bearer independent CS core network.

10.1.2.1 Call clearing from the network side

If the following release indication is received from the preceding/succeeding node, a corresponding procedure shall be performed.

1. If the release indication is a CANCEL request from the preceding node during call establishment, the MSC server shall send a 200 OK response for the CANCEL request and a 487 Request Terminated response for the initial INVITE request to the preceding node, and perform the procedure described below the bullet list.

2. If the MSC server receives a CANCEL request from the preceding node after having returned a final response to the initial INVITE request, the MSC server shall send a final response for the CANCEL request but no further action taken.

3. If the release indication is a BYE request from the preceding/succeeding node, the MSC server shall send a 200 OK response for the BYE request to the preceding/succeeding node, send a 487 Request Terminated response for the initial INVITE request to the preceding/succeeding node if no final response for the initial INVITE request has been sent out yet, and perform the procedure described below the bullet list.

4. If the request message is a failure response for the initial INVITE request for an establishing call from the succeeding node, the MSC server shall send an ACK for the failure response to the succeeding node, perform the procedure described below the bullet list.

In above cases, the MSC server shall perform a procedure to release any MGW allocated resources reserved for the call for the network side. If any resources were seized in the MGW for the network side, the MSC server shall use the Release RTP Termination procedures to indicate to the MGW to remove the bearer termination.

10.1.2.2 Call clearing to the UE

The MSC server initiates call clearing towards the UE and requests release of the associated radio resources as described in 3GPP TS 23.108[18]. Once the call clearing and the release of the associated radio resources have been completed, the MSC server releases any MGW allocated resources for the access side. If any resources were seized in the MGW, the MSC server uses the Release Termination procedure to requests the MGW to remove the access side bearer termination.

10.2 User Initiated

The user initiated call clearing shall be performed in accordance with 3GPP TS 23.108 [18]. Procedures used for user initiated call clearing are equivalent to those for a BICC-based Nc interface with the exception of the procedures described in the following clauses.

10.2.1 Call clearing from the UE

There is no specific requirement for the call clearing from the UE in the SIP-I based CS network.

10.2.2 Call clearing to the network side

The MSC server shall perform a corresponding procedure as below:

1. As the originating MSC server: If the MSC server has sent an initial INVITE request to the succeeding node and a non-100 provisional response or final response for the initial INVITE request has been received, the MSC server shall send a BYE request to the succeeding node to release the call. If only a 100 response for the initial INVITE request has been received, the MSC server shall send a CANCEL request to release the call. If no response for the INVITE request has been received, the MSC server shall wait for any case above being fulfilled.

2. As the terminating MSC server: If the MSC server received an initial INVITE request from the preceding node and no final response for the initial INVITE request has been sent out, the MSC server shall send an appropriate failure response for the initial INVITE request to the preceding node. If the MSC server received an initial INVITE request from the preceding node and a successful final response for the initial INVITE request has been sent out, the MSC server shall send a BYE request to release the call.

In above cases, the MSC server shall perform a procedure to release any MGW allocated resources for the network side when the MSC server receiving or sending a failure response for the initial INVITE request or sending a BYE request. If any resources were seized in the MGW, the MSC server shall use the Release RTP Termination procedures to indicate to the MGW to remove the incoming side bearer termination and that the bearer can be released towards the preceding MGW.

10.3 (G)MSC server Initiated

The following clauses describe the additional requirements for (G)MSC server initiated call clearing in the bearer independent CS core network.

10.3.1 GMSC server

10.3.1.1 Call clearing to the destination side

If the call is already established towards the destination, call clearing to the destination side from the GMSC server is performed as the originating MSC server described in clause 10.2.2.

10.3.1.2 Call clearing to the originating side

Call clearing to the originating side from the GMSC server is performed as the terminating MSC server described in clause 10.2.2.

10.3.2 MSC server

10.3.2.1 Call clearing to the UE

Call clearing to the UE is performed as described in clause 10.1.2.2.

10.3.2.2 Call clearing to the network side

Call clearing to the network side is performed as described in clause 10.2.2.

10.4 MGW Initiated

The following clauses describe the additional requirements for MGW initiated call clearing in the bearer independent CS core network.

10.4.1 GMSC server

10.4.1.1 Bearer released on the destination side

After the GMSC server received the Bearer Released procedure from the MGW on the destination side, call clearing to the destination side and the originating side from the GMSC server is performed as the originating MSC server and terminating MSC server described in clause 10.2.2 respectively.

10.4.1.2 Bearer released on the originating side

After the GMSC server received the Bearer Released procedure from the MGW on the originating side, call clearing to the originating side from the GMSC server is performed as the terminating MSC server described in clause 10.2.2. If the call is already established towards the destination side, call clearing to the destination side is performed as the originating MSC server described in clause 10.2.2.

10.4.2 MSC server

10.4.2.1 Bearer released on the access side

After the MSC server has received the Bearer Released procedure from the MGW on the access side, the MSC server shall release the access resources as described in clause 10.1.2.2. If the call is already established towards the network side, call clearing to the network side is performed as described in clause 10.2.2.

10.4.2.2 Bearer released on the network side

After the MSC server has received the Bearer Released procedure from the MGW on the network side, the MSC server shall clear the call to the network side as described in clause 10.2.2, and clear the call to the UE as described in clause 10.1.2.2.

10.5 Call Clearing for Iu Interface on IP

Procedures for Call Clearing where the Iu Interface is on IP are as described in clauses through 10.1 to 10.4, with the exception that only the Access side procedures apply and the Release Bearer procedure is not sent. For Iu Interface on IP, the Release RTP Termination procedures for IP are used to clear the MGW termination from the (G)MSC.

11 Handover/Relocation

NOTE: All message sequence charts in this clause are examples. All valid handover/relocation message sequences can be derived from the example message sequences and associated message pre-conditions.

11.1 UMTS to UMTS

In the context of the following clauses, the terms RNS or RNC refer also to a GERAN BSS or BSC (respectively) when serving an UE in Iu mode.

11.1.1 Intra-MSC SRNS/SBSS Relocation

The procedures specified in 3GPP TS 23.009 [65] for "Intra-3G_MSC SRNS Relocation" shall be followed. The following clauses describe the additional requirements for the bearer independent CS core network.

11.1.1.1 Intra-MGW Relocation

The handling of Intra-MSC Intra-MGW SRNS/SBSS Relocation is as defined in the clause 8.1.1 of 3GPP TS 23.205 [2].

11.1.1.2 Inter-MGW Relocation

Procedures used for the Intra-MSC Inter-MGW SRNS/SBSS Relocation are equivalent to those for a BICC-based Nc interface with the exception that the procedures for establishing the bearer between the MGW will be in accordance with normal procedures as defined for SIP-I (assuming that the bearer is in accordance with clause 5.4).

11.1.2 Basic Inter-MSC SRNS/SBSS Relocation

Procedures used for the basic Inter-MSC SRNS/SBSS Relocation are equivalent to those for a BICC-based Nc interface with the exception that the call and bearer establishment procedures between the MSCs and MGWs will be in accordance with normal procedures as defined for SIP-I (assuming that the bearer is in accordance with clause 5.4).

11.1.3 Subsequent Inter-MSC SRNS/SBSS Relocation back to the Anchor MSC

Procedures used for the subsequent Inter-MSC SRNS/SBSS Relocation back to the anchor MSC are equivalent to those for a BICC-based Nc interface with the exception that the call clearing and bearer release procedures between the MSCs and MGWs will be in accordance with normal procedures as defined for SIP-I.

11.1.4 Subsequent Inter-MSC SRNS/SBSS Relocation to a third MSC

Procedures used for the subsequent Inter-MSC SRNS/SBSS Relocation to a third MSC are equivalent to those for a BICC-based Nc interface with the exception that the call establishment, call clearing, bearer establishment and bearer release procedures between the MSCs and MGWs will be in accordance with normal procedures as defined for SIP-I (assuming that the bearer is in accordance with clause 5.4).

11.1.5 SRNS/SBSS Relocation with Iu on IP

If IuCS on IP is supported by the MSC server, the Core Network side procedures described in clauses through 11.1.1 to 11.1.4 shall apply, and the Access Network side procedures described in clause 8.1.5 of 3GPP TS 23.205 [2] shall apply.

11.2 UMTS to GSM

Intra-MSC SRNS/SBSS Relocation, Basic Inter-MSC SRNS/SBSS Relocation, Subsequent Inter-MSC SRNS/SBSS Relocation back to the Anchor MSC, Subsequent Inter-MSC SRNS/SBSS Relocation to a third MSC procedures shall be defined respectively. SRNS/SBSS Relocation with Iu on IP shall also be defined.

11.3 GSM to UMTS

Intra-MSC SRNS/SBSS Relocation, Basic Inter-MSC SRNS/SBSS Relocation, Subsequent Inter-MSC SRNS/SBSS Relocation back to the Anchor MSC, Subsequent Inter-MSC SRNS/SBSS Relocation to a third MSC procedures shall be defined respectively. SRNS/SBSS Relocation with Iu on IP shall also be defined.

11.4 GSM to GSM

Intra-MSC SRNS/SBSS Relocation, Basic Inter-MSC SRNS/SBSS Relocation, Subsequent Inter-MSC SRNS/SBSS Relocation back to the Anchor MSC, Subsequent Inter-MSC SRNS/SBSS Relocation to a third MSC procedures shall be defined respectively. SRNS/SBSS Relocation with Iu on IP shall also be defined.

12 Impact to existing services

12.1 Supplementary Services

12.1.1 Explicit Call Transfer (ECT)

Party B is the subscriber controlling the Explicit Call Transfer Call (served mobile subscriber). Party A is the first remote called party (held party). Party C is the second remote called party. MSC-A, MSC-B and MSC-C is the MSC server served for Party A, Party B and Party C respectively.

After a call between Party A with Party B has been established successfully and Party A is held, a new call between party B with Party C is established successfully. After getting the ECT request, MSC-B initiates the ECT service procedure by changing through the connection between Party A with Party C.

For the Explicit Call Transfer in SIP-I based Nc interface, two possible approaches are considered:

1. After receiving an ECT request from Party B UE, MSC-B sends a REFER request to MSC-A with Refer-To header with the identifier of Party C and the dialog ID with the call between Party B and Party C to request MSC-A place a new call to Party C with replace the ongoing call between Party B and Party C. Additionally, MSC-B sends ECT notifications to MSC-A and MSC-C. Figure 12.1.1.1 is an example service flow diagram for this approach.

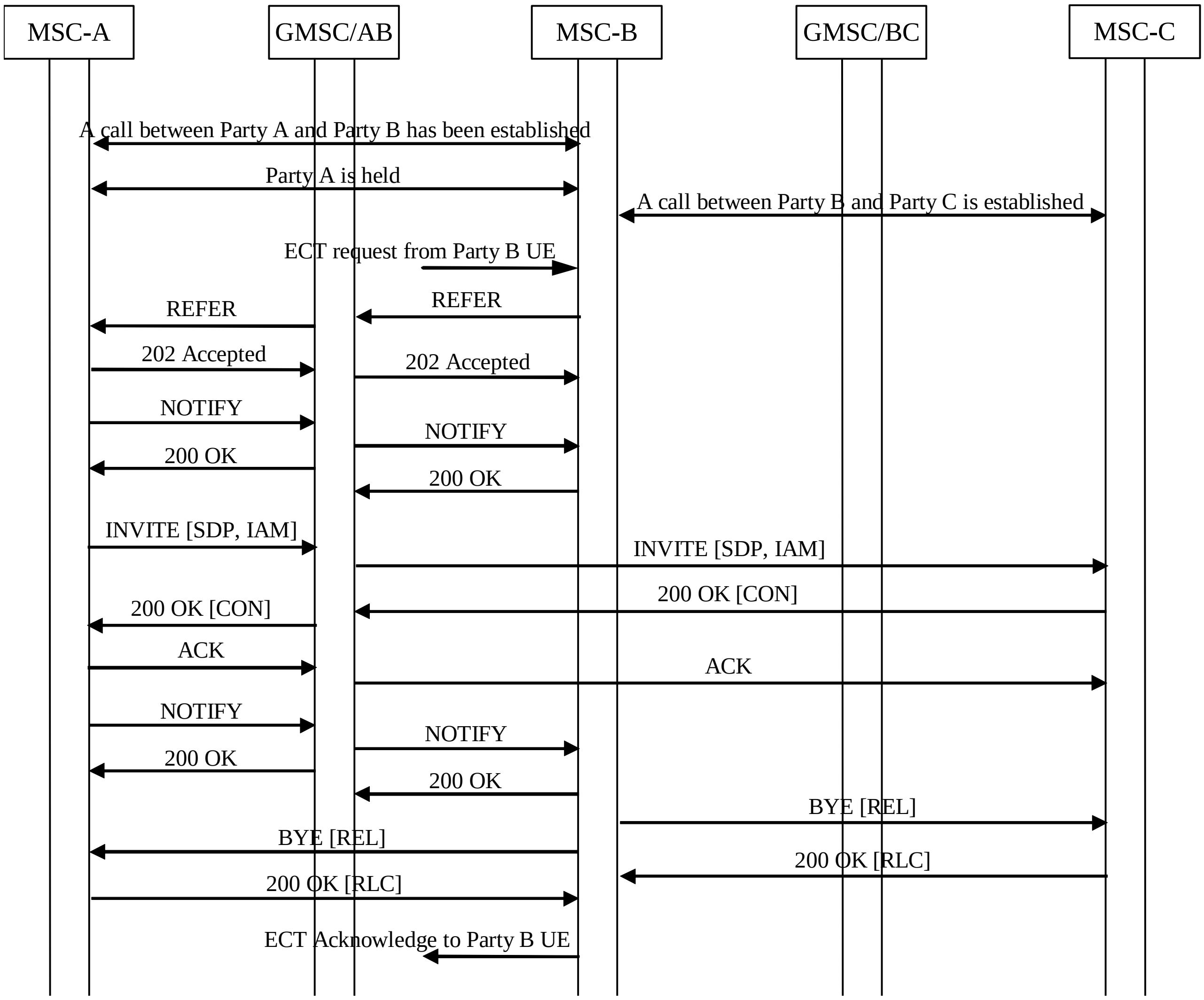

Figure 12.1.1.1: ECT service flow diagram

As this approach, it takes advantages of MSC-B quitting from the call between Party A and Party C in principle. And resources of the initial calls traversed will be freed from the new call.

When the new initial INVITE message goes to Party C, the prior call in Party C will be replaced to avoiding call rejecting of busy and rendering of a new call attempt, and the REFER request to request initiating the new call between Party A and Party C will take the SIP dialog ID of the call between Party B and Party C. However, as the GMSC works as a B2BUA, MSC-B will be unable to get the ID of the dialog between MSC-C and GMSC/BC. Since there is no mechanism to ensure the new initial INVITE message goes to Party C via GMSC/BC, the new call establishment and the ECT service will fail if this approach is used.

2. After receiving an ECT request from Party B UE, MSC-B shall retrieve Party A from hold by using a re-INVITE request and through-connection the bearer terminations between Party A and Party C. Additionally, MSC-B sends ECT notifications to MSC-A and MSC-C using INFO requests. Figure 12.1.1.2 is an example service flow diagram for this approach.

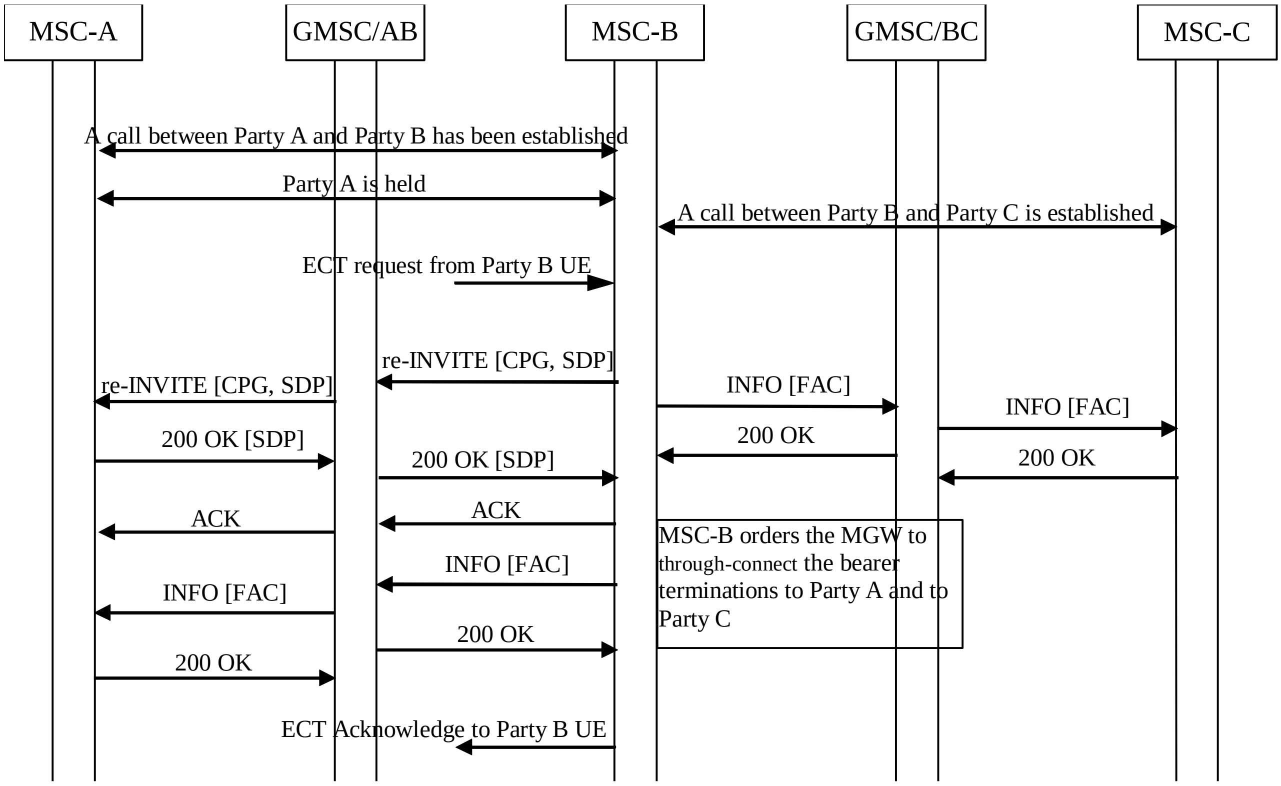

Figure 12.1.1.2: ECT service flow diagram

As approach 2, MSC-B will always be in the call between Party A and Party C. As an optimization, MSC-B may remove the MGW under its control to make a direct bearer interworking between Party A and Party C to reduce the delay of bearer packets, increase the voice quality and save network resources.

Alternatively, after receiving an ECT request from Party B UE, MSC-B shall retrieve Party A from hold and may align the codecs between the connected parties at MSC-A and MSC-C by using a re-INVITE requests and through-connection the bearer terminations between Party A and Party C as shown in Figure 12.1.1.3. Additionally, MSC-B sends ECT notifications to MSC-A and MSC-C using INFO requests.

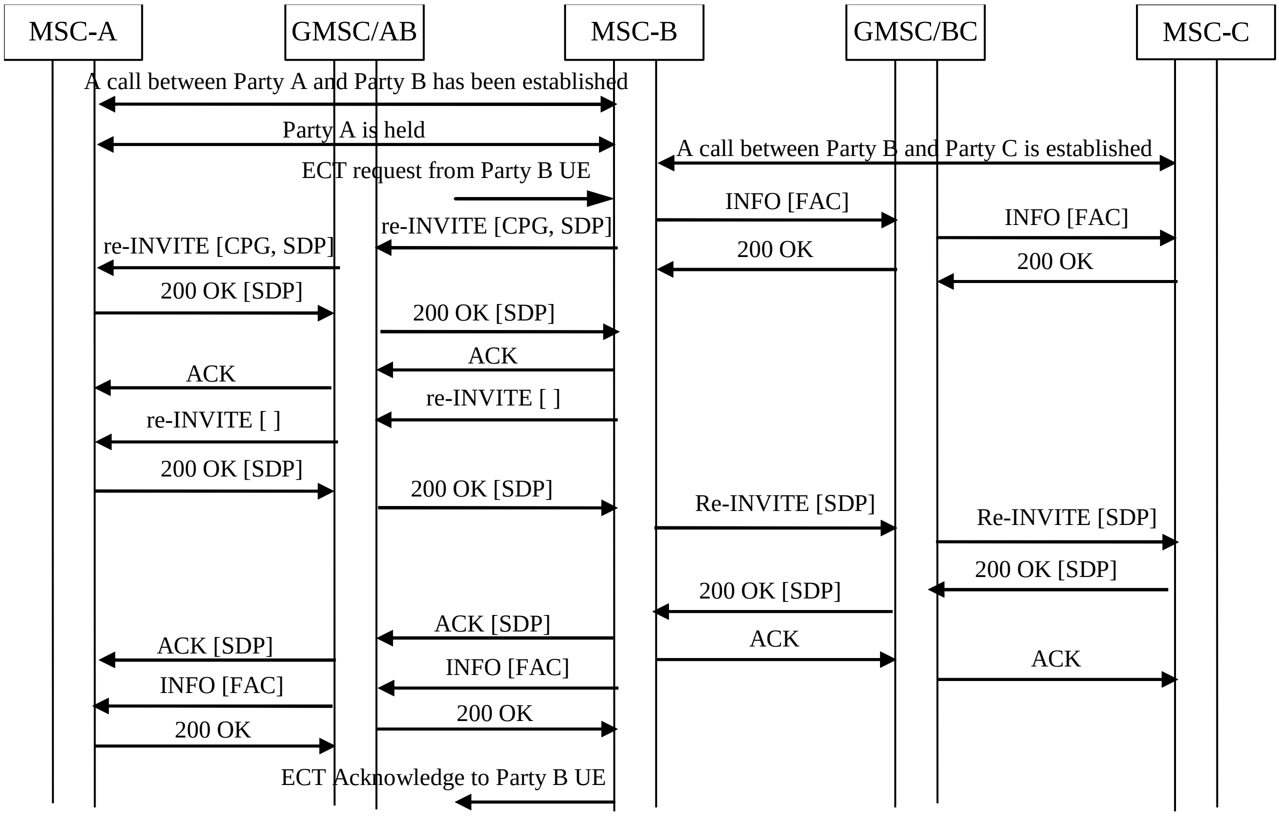

Figure 12.1.1.3: ECT service flow diagram

As a variation of the call flow shown in Figure 12.1.1.3, if the subscriber at MSC-C has answered, it is possible combine the codec alignment and retrieval options in one end-to-end offer/answer exchange as shown in Figure 12.1.1.4.

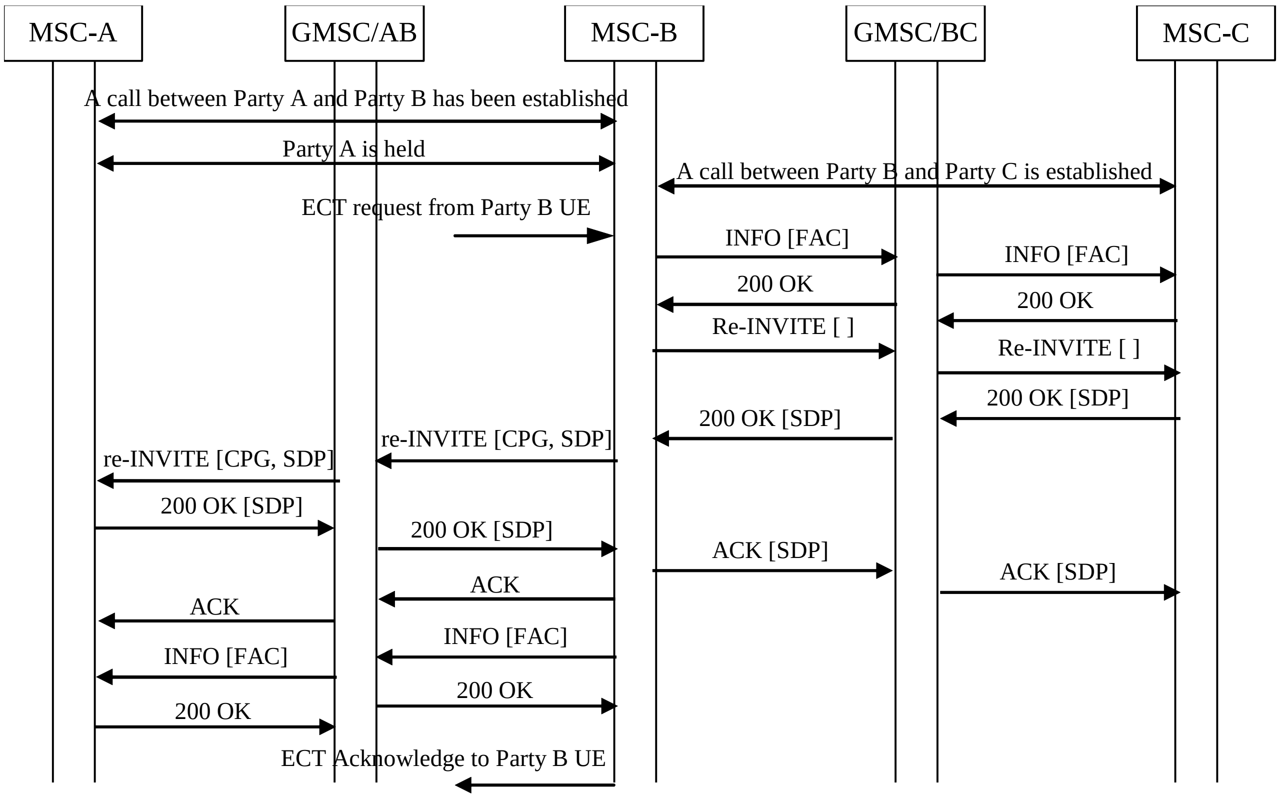

Figure 12.1.1.4: ECT service flow diagram

Of the two approaches only approach 2 does not result in a call failure scenario, therefore approach 2 shall be supported as the approach of Explicit Call Transfer service and the REFER method shall not be used in the Explicit Call Transfer procedure..

The procedures specified in 3GPP TS 23.091 [44] for the Explicit Call Transfer supplementary service shall be followed. The following clauses describe the additional requirements for the SIP based CS core network.

12.1.1.1 Connection of remote parties

If the result of the ECT checks is successful the MSC server will order the MGW to connect the bearer termination of the Party C to the bearer termination of the Party A. As a result of this action the held party will be retrieved.

If the call towards the Party C has not been answered, the MSC server requests the MGW to both-way through-connect the bearer termination towards the Party C.

12.1.1.2 IU/A-interface release

The served party is disconnected after a successful transfer request. The call towards the served mobile subscriber shall be released as described in the clause for call clearing.

12.1.1.3 Failure handling in MSC server

If the bearer terminations for the remote parties can not be connected successfully, the MSC server shall reject the ECT request.

12.2 Number Portability

Number portability refers to the ability of a mobile subscriber to change mobile network subscription within a portability domain whilst retaining his/her original MSISDN(s).

There are two possible approaches to facilitate this feature in SIP-I based networks:

- Option 1: Number portability information is contained within the encapsulated ISUP IAM message
- Option 2: Support of rn and npdi parameters defined in IETF RFC 4694 [72]

With Option 1, the routeing number (RN) and MSISDN of the called party are carried within the encapsulated ISUP IAM (according to agreements between networks within a portability domain) and these are used by the GMSC to route the call to the GMSC of the next network (using either the IN-based or MNP-SRF (call-related) solution as specified in 3GPP TS 23.066 [73]).

With Option 2, the IETF RFC 4694 [72] defines a set of parameters to enable the support of number portability within "tel" URI. These parameters can be passed to the next network node after the number portability database dip has been performed (using either the IN-based or MNP-SRF (call-related) solution as specified in 3GPP TS 23.066 [73]).

It is recommended that of the two approaches, Option 1 shall be used on the Nc interface while the number portability parameters (rn and npdi) as defined in IETF RFC 4694 [72] may be supported as an option for rendering MNP when inter-working with external SIP-I networks via an IWF.

12.3 SCUDIF

SCUDIF applies a transparent transport of the Multimedia payload with a 64kbit. Therefore, the existing CLEARMODE IETF RFC 4040 [34] is well suited as user plane transport format for multimedia. If the speech service is used within SCUDIF, ordinary speech coding is applied.

There is no way to signal the use of MUME within SDP with existing IETF specifications. Signalling extensions will be required to introduce the MUME codec into the SDP codec negotiation for call setup and support for mid-call switching between speech and multimedia modes. The following possible encodings for the MUME codec have been proposed:

- A revision of the CLEARMODE IETF RFC 4040 [34] within IETF to introduce a MIME parameter to express that the CLEARMODE codec is used to transport multimedia. If this extension is done, IETF RFC 4040 [34] could also be expanded to allow signalling within SDP of other CLEARMODE applications such as Nx64, V.120 and V.110. Such CS data call related possible extensions are further discussed in Clause 11.5. SCUDIF related additions could be included in a bis version of IETF RFC 4040 [34]. Older CLEARMODE terminals receiving the new MIME parameters of the CLEARMODE MIME type would ignore these parameters and may therefore not be aware of the Multimedia payload within CLEARMODE. On the one hand a multimedia call setup may then still succeed if the terminal assumes the right payload within CLEARMODE. On the other hand, the call setup would fail if the terminal assumes some wrong payload.

- As an alternative, 3GPP could define an own RTP payload type with the same format as IETF RFC 4040 [34], but separate MIME registration. A 3GPP defined payload type would have the advantage to be available earlier, but bears the risk of a more limited support outside 3GPP, unless an interworking to explicit video codec signalling is performed at the edge of the network. Older CLEARMODE terminals not recognising the new MIME type would reject this codec, thus causing a fallback to speech.

BICC based SCUDIF is only recognized within 3GPP CS networks, but it is desirable to achieve some interoperability towards external SIP based networks. To achieve interworking of SCUDIF towards non-3GPP networks attached to the SIP-I network a similar interworking to SIP-I as being developed for CS-to-IMS interworking of the service in TR 29.863 [45] could be used. Explicit video codecs within SDP would be offered towards the external network and an interworking node would modify the payload and interwork the inband H.245 codec negotiation with the external SIP codec negotiation. As this approach requires a costly interworking and is not required in all scenarios, it should be optional.

In another scenario, an external SIP-I network is used as a transit between two 3GPP CS domains. In this scenario, using the CLEARMODE codec avoids this interworking and guarantees maximal call success rate with minimum delay, as it allows for an end-to-end inband H.245 codec negotiation. If the Interworking node does not know the destination network of a MO call, it may offer both encoding possibilities, as discussed within TR 29.863 [45].

12.4 GTT

On the Nb interface, GTT is carried either as CTM in audio, V.18 tones in TDM, or as T.140 text within a CS multimedia bearer. CTM is only used on Nb when the peer endpoint is known by an established convention to support CTM, since there is no way to signal its use in the network. In IMS, GTT is carried as T.140 text according to 3GPP TS 26.235 [33].

12.5 Impacts on Data Call Handling

The transport format of data calls in a SIP-I network at the network side of the interworking function at the A-MSC shall be in accordance with the Table 12.5.1 below for data calls originating in the 3GPP CS domain.

For incoming data calls from external networks, the transport format used in the external network needs to be supported. It is expected that the transport format of many incoming calls is also as listed in Table 12.5.1, as this table has been derived from Table 26 in ITU-T Q.1912.5 [16], although that table is not mandated for a SIP-I network according to ITU-T Q.1912.5.

In many cases, the CLEARMODE codec, IETF RFC 4040 [34], is used as transport format.

A SIP-I 3GPP Cs Domain shall support in-band Fax/modem call via the VBD codec defined in V.152 [50]. For fax calls T.38 can be used alternatively to the VBD codec.

Table 12.5.1- Coding of TMR/USI/HLC from SDP: SIP to ISUP

m= line			b= line (NOTE 4)	a= line	TMR parameter	USI parameter (optional) (NOTE 1)		HLC parameter (optional)
<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value> (NOTE 5)	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>	TMR codes	Information Transport Capability	User Information Layer 1 Protocol Indicator	High Layer Characteristics Identification
audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A	"3.1KHz audio"			(NOTE 3)
audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMU/8000	"3.1KHz audio"			(NOTE 3)
audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	*"3.1KHz audio"*			(NOTE 3)
audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000	*"3.1KHz audio"*			(NOTE 3)
audio	RTP/AVP	9	AS: 64 kbit/s	rtpmap:9 G722/8000	*"64 kbit/s unrestricted"*	*"Unrestricted digital inf. w/tones/ann"*		
audio	RTP/AVP	Dynamic PT	AS: 64 kbit/s	rtpmap:<dynamic-PT> CLEARMODE/8000 (NOTE 2)	*"64 kbit/s unrestricted"*	*"Unrestricted digital information"*		
image	udptl	t38	N/A or up to 64 kbit/s	Based on T.38 [28]	*"3.1 KHz audio"*			
image	tcptl	t38	N/A or up to 64 kbit/s	Based on T.38 [28]	*"3.1 KHz audio"*			

NOTE 1 In this table the codec G.711 is used only as an example. Other codecs are possible.
NOTE 2 CLEARMODE is specified in RFC4040 [34].
NOTE 3 HLC is normally absent in this case. It is possible for HLC to be present with the value "Telephony", although 6.3.1/Q.939 indicates that this would normally be accompanied by a value of "Speech" for the Information Transfer Capability element.
NOTE 4 If the b=line indicates a bandwidth greater than 64kbit/s then the call may use compression techniques or reject the call with a 415 response indicating that only one media stream of 64kbit/s is supported.
NOTE 5 <bandwidth value> for <modifier> of AS is in units of kbit/s.

According to the current principles in 3GPP specification TS 23.202 [47] and TS 29.007 [48], a data call is identified using the ISUP TMR and USI parameters. Information elements derived from these parameters are also applied in MAP and in signalling towards the UE. To keep impacts to existing implementations minimal, it is desirable that the TMR and USI parameters in the encapsulated ISUP are also used in SIP-I based CS domain to identify data calls.

In addition or as an alternative, the CLEARMODE codec could be extended to allow signalling within SDP of other CLEARMODE applications such as Nx64, V.120 and V.110. However, such an extension is currently not used in external SIP-I network and therefore may lead to interworking problems.

In some scenarios, the incoming TMR/USI information is not sufficient to identify a MT data call and signalling interactions with the terminating UE are then used to determine the type of the data call. To enable such interactions, it is beneficial if several formats, e.g. a speech codec plus CLEARMODE, are offered by the originating side within SDP. The VBD and/or the T.38 codec should also be offered both if the call is known to be a fax/modem call or when it cannot be determined from the signalling if the call is a fax/modem or a speech call, for example in case of calls of PSTN origin.

NOTE: It is FFS if the terminating side could select G.711 as transport format for data calls if no other suitable format has been offered.

NOTE: Interactions between TMR/USI information in the encapsulated ISUP and the SDP codec negotiation as described in the present specification require further description.

The transport format of many data calls is modified by the interworking function. At the access side of the interworking function, a 3GPP CS Domain core network connection may be encountered after an inter-MSC handover.

NOTE: The core network transport format at the access side of he interworking function is FFS, e.g. for non transparent facsimile group 3 calls according to 3GPP TS 23.146 [49].

12.6 Support of DTMF

SIP-I does not include a method of signalling DTMF events in the manner provided by BICC. Instead SIP-I transports DTMF within the RTP telephone-event IETF RFC 4733 [35] if the RTP telephony event payload type is negotiated between the peer SIP-I entities; alternatively SIP-I may also transport the DTMF in-band ("within the codec") when the selected codec allows transparent DTMF transport.

SIP-I based Nc shall support DTMF transport via RTP telephone-events, which is the recommended DTMF transport on SIP-I based Nc. However, SIP-I based Nc may also support in-band DTMF transport for open interoperability with external SIP-I network. Depending on whether the MSC Server has configured or not the RTP telephone-event payload at the MGW, the MGW shall detect or transmit DTMF within the RTP payload type according to RFC 4733 [35] or in-band ("within the codec") if supported.

When interfacing to the PLMN using ISUP, the MSC receiving DTMF events from a mobile station signals those events to its MGW so that the MGW can generate the DTMF tones in-band within the bearer plane to the PLMN. When interfacing to a SIP-I network, the MSC receiving DTMF events from a mobile station signals those events to its MGW so that the MGW shall signal the DTMF digits in the bearer plane according to the DTMF transport mode configured.

When interworking DTMF events received from a SIP-I network to BICC, the MGW shall detect the DTMF events in the bearer plane according to the DTMF transport mode configured and indicate them to its controller for carriage in the BICC signalling plane. When interworking DTMF events received from a BICC network to SIP-I, the MGW controller shall signal DTMF events received in the signalling plane to its MGW for transmission in the bearer plane towards the SIP-I network according to the DTMF transport mode configured.

When interworking to a SIP-I network, in-band DTMF received in the bearer plane from a BICC or ISUP network shall be detected at the MGW and translated to the corresponding telephone-events for carriage in the bearer plane of the SIP-I network if DTMF transport via RTP telephone-event has been configured. Otherwise, in-band DTMF shall be sent in-band.

12.7 Impacts on Call Deflection Service

The section is to analyse possible impacts on Call Deflect service handling compared to the current handling based upon BICC or ISUP.

12.8 Impacts on Call Forwarding Services

The section is to analyse possible impacts on Call Forwarding services handling compared to the current handling based upon BICC or ISUP.

12.9 Impacts on Call Hold

The section is to analyse possible impacts on Call Hold handling compared to the current handling based upon BICC or ISUP.

12.10 Impacts on GSM Fax

The section is to analyse possible impacts on GSM Fax handling compared to the current handling based upon BICC or ISUP.

12.11 Impacts on Bearer Redirection

The section is to analyse possible impacts on Bearer Redirection handling compared to the current handling based upon BICC or ISUP.

13 Interworking between SIP-I based 3GPP CS Domain and BICC based 3GPP CS Domain

3GPP currently standardises the interworking between 3GPP CS domain with BICC or ISUP as signalling protocol and external SIP-I networks in TS 29.164 [46]. It is expected that similar procedures are applicable for the Interworking between a SIP-I based 3GPP CS Domain and BICC based 3GPP CS Domain. The scope of TS 29.164 could be extended to cover also this scenario.

Technical differences could arise if SIP-I was applied in a different manner on the Nc interface and within external networks. To limit such differences, it is desirable to allow for a similar usage of SIP-I on the Nc interface as within external networks.

14 Interworking between SIP-I based 3GPP CS Domain and 3GPP IP Multimedia Subsystem

3GPP standardises the interworking between 3GPP CS domain with BICC or ISUP as signalling protocol and 3GPP IP Multimedia (IM) Core Network (CN) in 3GPP TS 29.163 [25].

3GPP also standardises the interworking between 3GPP CS domain with BICC or ISUP as signalling protocol and external SIP-I networks in 3GPP TS 29.164 [46]. Clause 13 states that it is desirable to allow a similar usage of SIP-I on the Nc interface as that used by external SIP-I networks in order to enable the interworking between a 3GPP SIP-I based Nc interface and a 3GPP CS domain with BICC or ISUP as the signalling protocol.

The standardisation of these interworking scenarios allows for the interworking between a SIP-I based 3GPP CS Domain and a 3GPP IP Multimedia Subsystem. Figures 14.1 and 14.2 detail this interworking scenario.

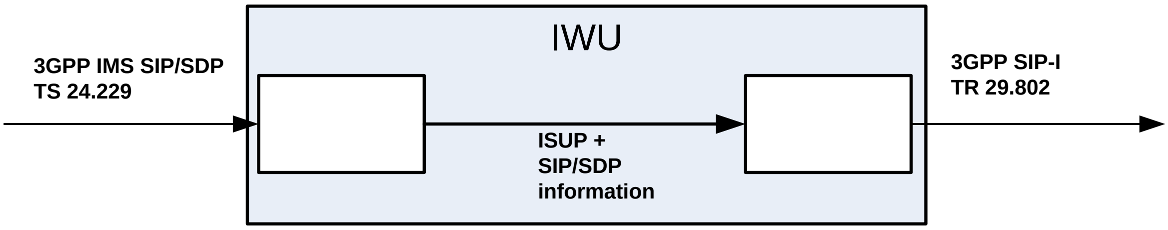

Figure 14.1: Interworking from 3GPP IMS to 3GPP SIP-I base Nc Interface

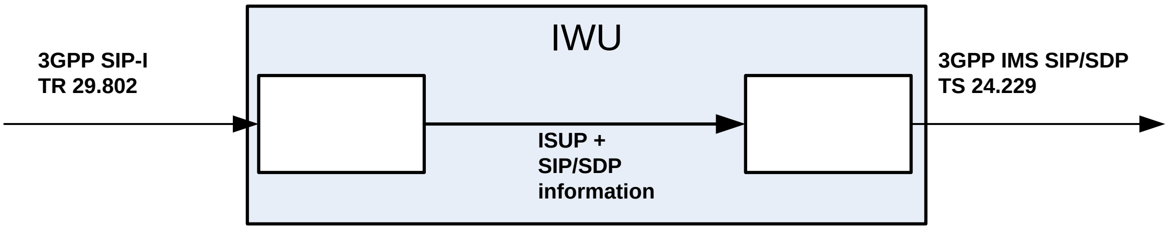

Figure 14.2: Interworking from 3GPP SIP-I base Nc Interface to 3GPP IMS

In both Figure 14.1 and Figure 14.2 it should be clearly understood that within the IWU the ISUP information and SIP/SDP information may be utilised to in order to interwork the call control protocol.

The IWU may be implemented as SIP B2BUA, utilising the SIP/SDP received in the received request to generate the new request. ISUP information encapsulated in messages received from the SIP-I based CS domain are de-encapsulated and information related to specific services is transformed to corresponding SIP extensions used within the IMS. The ISUP is not encapsulated in the messages sent towards the IMS. Information related to specific services within SIP extensions in messages received from the IMS side may be transformed to ISUP information generated according to the procedures in TS 29.163 [25], which is then encapsulated in the SIP-I messages sent towards the SIP-I based CS domain. The encapsulation of the ISUP into the SIP-I messages may be done according to the procedures in TS 29.164 [46].

The control plane between the IM CN Subsystem supporting SIP and a 3GPP CS network supporting a SIP-I based Nc interface is as shown in Figure 14.3.

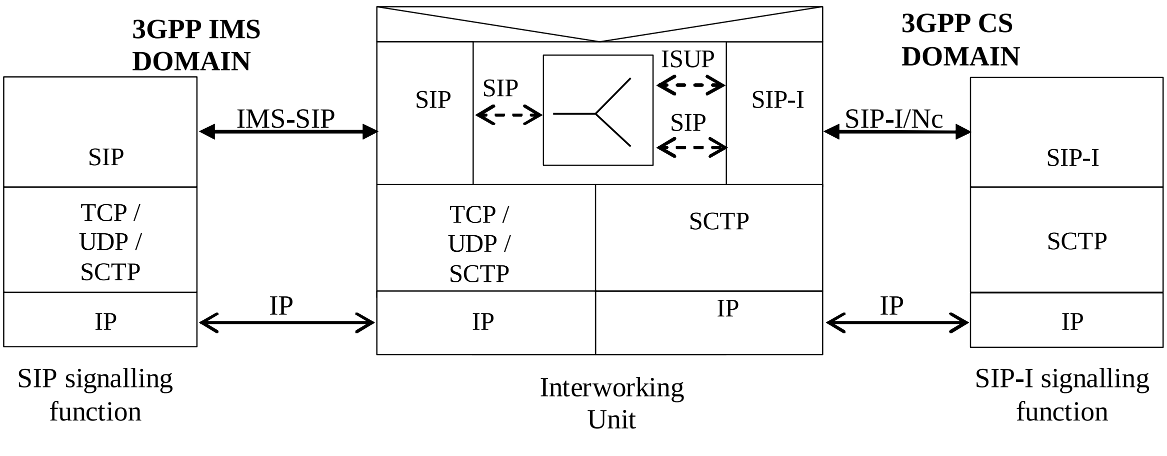

Figure 14.3: Control plane interworking between the IM CN subsystem and a 3GPP CS network supporting SIP-I based Nc interface

15 Impacts to Messages/Procedures and their contents

15.1 Impacts to Messages between (G)MSC servers

A 3GPP node supporting the SIP-I based Nc interface shall support the methods defined as mandatory according to clause 5.2, and may support the methods defined as optional according to clause 5.2.

The procedural definition and contents (e.g. SIP headers, embedded ISUP, SDP) shall be detailed within Stage 2 and Stage 3 specifications.

15.2 Impacts to Procedures between (G)MSC server and MGW

The procedures in Table 15.2.1 are defined for the purpose of the (G)MSC-S interaction with MGW in the SIP-I based CS network.

Table 15.2.1: Procedures used between (G)MSC-S and MGW in the SIP-I based CS network

Procedure Name	Procedure Description	Corresponding Procedure
Reserve RTP Connection Point	This procedure is used to reserve local connection addresses and local resources.	Reserve IMS connection point in the Mn interface as defined in TS 29.332 [20].
Configure RTP Resources	This procedure is used to select media-processing resources for a Nb interface connection.	Configure IMS resources in the Mn interface as defined in TS 29.332 [20]..
Reserve RTP Connection Point and Configure Remote Resources	This procedure is used to reserve media-processing resources for an Nb interface connection.	Reserve IMS Connection point and configure remote resources in the Mn interface as defined in TS 29.332 [20].
Release RTP Termination	This procedure is used by the (G)MSC-S to release a termination towards the remote MGW and free all related resources.	Release IMS termination in the Mn interface as defined in TS 29.332 [20].
Detect RTP Tel Event	This procedure is used by the (G)MSC-S to request the MGW the detection of telephony events signalled within RTP according to RFC 2833bis [35] and the notification of received telephony events.	Detect IMS RTP Tel event in the Mn interface as defined in TS 29.332 [20].
Notify RTP Tel Event	This procedure is used by the MGW to notify the (G)MSC-S about the detection of telephony events signalled within RTP according to RFC 2833bis [35].	Notify IMS RTP Tel event in the Mn interface as defined in TS 29.332 [20].
Send RTP Tel Event	This procedure is used by the (G)MSC-S to request from the MGW to signal a telephone event within RTP according to RFC 2833bis [35].	Send IMS RTP Tel event in the Mn interface as defined in TS 29.332 [20].
Stop RTP Tel Event	This procedure is used by the (G)MSC-S to request the MGW to stop signalling a telephone event within RTP according to RFC 2833bis [35].	Stop IMS RTP Tel event in the Mn interface as defined in TS 29.332 [20].
Bearer Released	This procedure is used by the MGW to indicate towards the (G)MSC-S that an error occured on an termination which requires the release of the termination.	IMS Bearer Released in the Mn interface as defined in TS 29.332 [20].
Change RTP Through-Connection	This procedure is used by the (G)MSC-S to request the MGW to through connect the terminations.	Change IMS ThroughConnection in the Mn interface as defined in TS 29.332 [20].
Send RTP Tone	This procedure is used by the (G)MSC-S to request the MGW to send tone.	IMS Send Tone in the Mn interface as defined in TS 29.332 [20].
Stop RTP Tone	This procedure is used by the (G)MSC-S to request the MGW to stop tone.	IMS Stop Tone in the Mn interface as defined in TS 29.332 [20].
RTP Tone Completed	This procedure is used by the MGW to indicate the (G)MSC-S that a tone has finished being generated at a termination.	IMS Tone Completed in the Mn interface as defined in TS 29.332 [20].

16 Conclusion and Recommendations

Through the preceding technical investigation within this technical report, it is concluded that providing a SIP-I based Nc interface as an alternative to the existing BICC definition is a feasible option.

It is recommended to 3GPP that the necessary detail within this report be used as a basis for further technical work within the Release 8 timeframe. It is further recommended that Stage 2 and Stage 3 work be specified within both existing specifications and in the creation of new specifications as defined within Annex A.

Annex A (informative): Impacts to Existing Specifications

Table A.1 identifies the existing specifications that require modification to define a SIP-I based Nc interface.

Table A.1

Existing Specification	Responsible WG	Brief summary of impacts
3GPP TS 23.205 [2]	CT4	Modify content to reflect that content is normative if using a BICC-based Nc interface. Addition of reference to a new Stage 2 specification for a SIP-I based Nc Interface.
3GPP TS 29.232 [10]	CT4	Addition of new procedures required on the Mc interface to support an RTP-based bearer on the Nb interface. The relevant procedures are identified within clause 15.2, and shall reference the procedures as defined by the Mn interface, 3GPP TS 29.332 [20].
3GPP TS 29.163 [25]	CT3	Include 3GPP SIP-I as a valid protocol for interworking between IMS and CS networks.
3GPP TS 23.153 [17]	CT4	Modify content to allow for a 3GPP SIP-I based codec negotiation with RTP-based bearer.

Table A.2 identifies the new specifications that are required to define a SIP-I based Nc interface.

Table A.2

New Specification	Responsible WG	Brief summary of impacts
3GPP TS 23.aaa	CT4	Stage 2 normative content for a SIP-I based Nc Interface. Shall contain call flows and scenarios.
3GPP TS 29.bbb	CT4	Stage 3 SIP-I based Nc profile definition. Shall also contain the codec negotiation mechanisms with reference to 3GPP TS 23.153 [17] procedures.

Annex B (informative): Change History

Change history							
Date	TSG #	TSG Doc.	CR	Rev	Subject/Comment	Old	New
06-2007	CT#36	CP-070337			V7.0.0 approved in CT#36	2.0.0	7.0.0