Technical Specification of the SIP (Gm) interface between the User Equipment (UE) and the NGN platform of Deutsche Telekom

1 TR 114

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Amendment 4.1: Support of GRUU, Bandwidth Reservation and Network Based Services

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1 Scope

This Amendment is an addition to 1 TR 114 V3.0.0.

2 References

See 1TR114 and 1TR114 Annex B_V020000_TS24229-b60.pdf

These are the relevant References out of 1TR114 Annex B_V020000_TS24229-b60:

[92] RFC 5626 (October 2009): "Managing Client Initiated Connections in the Session Initiation Protocol (SIP)".

[93] RFC 5627 (October 2009): "Obtaining and Using Globally Routable User Agent URIs (GRUUs) in the Session Initiation Protocol (SIP)".

3 Modifications to 1 TR 114

3.1 General

In configurations where bandwidth reservation and the network based CH, CW, Toggel and CONF services are supported by the IAD, the Support of GRUU as described in the following sections shall apply. The bandwidth reservation allows that for each end device connected to the IAD one active and one hold Session can be supported. The maximum of possible parallel active session is depended on the number of subscribed parallel calls within the network.

Note: This is depended on the contract between Deutsche Telekom and customer. Contractual issues are out of scope of this specification.

3.2 Definitions

SIP UA: This is the SIP UA acting as contact point towards the IMS Gm. The SIP UA is responsible for registering the IMPU's at the IMS (i.e. Registrar). Depended on implementation within the IAD the SIP UA may act as internal SIP registrar and forking Proxy against the port number/IMPU combination as defined below.

IAD SIP UA: A IAD SIP UA is a naming within this document to reflect the combination of IMPU and Port number. This IAD SIP UA may register at the SIP UA "virtually" or "real". How this procedure is done is due to the implementation. Dependent on implementation the IAD SIP UA may be the same function as the SIP UA.

Port Number: Port number defined for this Amendment is a unique identifier which is used for a physical or virtual interface to connect a SIP UA with an End Device.

GRUU: Global Routable URI. Within the scope of this document a GRUU shall be generated for each IAD SIP UA i.e the combination of contacted end device/ Port Number. For this document a self made GRUU as defined in RFC 5627 [53] shall apply.

End Device:

An End Device seen from IAD perspective could be:

1. A POTS phone, i.e. each analogue end device connected has its port number.
2. ISDN phone or terminal adapter. Depending on the implementation each ISDN Phone may have a port number or equivalent. The ISDN Adapter may have two virtual port number per S0 channel supported which may be dynamically provided. This is implementation depended.
3. DECT phone. For each DECT phone a Port Number.

4. Or any other device used for telephony which interacts with an IAD SIP UA or is acting as IAD SIP UA or UA.

3.3 Support of GRUU within the IAD

The following figure shows an possible implementation of an IAD:

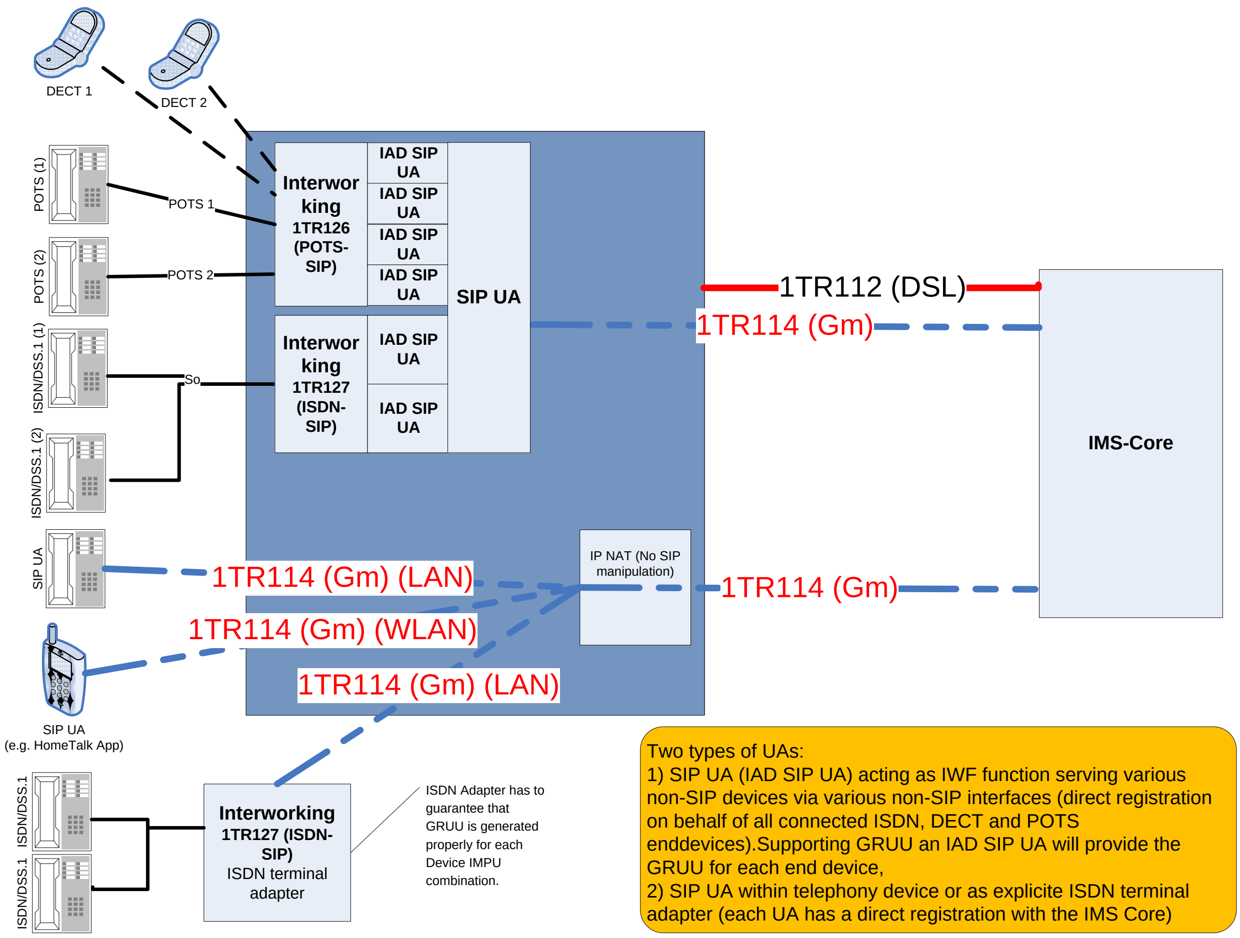

Figure 1 Possible architecture of an IAD with support of GRUU

For each IAD SIP UA the following procedures SHALL apply:

- Create an unique identifier (GRUU) for each IAD SIP UA,
- Depended on implementation SIP UA shall act as a registrar. It shall accept REGISTER from IAD SIP UA and create an unique identifier for each binding, or
- Generate for each IAD SIP UA a unique binding,
- For all requests and responses containing a SIP Contact header field (except REGISTER) sent by the SIP UA, the SIP UA shall add a GRUU to the SIP Contact header field based on the related IAD SIP UA.
- The Contact header with GRUU shall apply to the format as defined within RFC 5627 [93] as follows:
    - Contact: <contact-URI; gr=urn:uuid:**random-string**>

  – Example: Contact: <sip:iad-1@62.57.4.12; gr=kjh29x97us97d>

- The SIP UA shall extract telephone number from the P-Called-Party-Identity header and ring ports with matching configuration
- SIP UA shall create IMPU of configured telephone number and domain name and apply it to the P-Preferred-Identity and From headers

The GRUU shall be supported by the SIP Agents used within the IAD as follows.

Each IAD SIP UA shall have an internally registered GRUU as defined within RFC 5627 [93] which shall be populated within each contact of non-REGISTER requests and it’s responses. The use of Session-ID as defined in RFC 5626 [92] SHALL NOT apply.

The following procedure for constructing a self-made GRUU in RFC5227 [93] section 4.3 shall apply

- A self-made GRUU is one whose domain part equals the IP address or hostname of the user agent. The user part of the SIP URI is chosen arbitrarily by the user agent. Like all other GRUUs, the URI MUST contain the "gr" URI parameter, with or without a value, indicating it is a GRUU.

  Example: sip:alice@example.com;gr=kjh29x97us97d✓

As defined in RFC 5627 [93] the SIP UA shall us it’s own GRUUs as defined in Section 4.4 “Using One's Own GRUUs” of RFC5627 [93] as follows:

> A SIP UA SHOULD use a GRUU when populating the Contact header field of dialog-forming and target refresh requests and responses. In other words, a UA compliant to this specification SHOULD use one of its GRUUs as its remote target. This includes:
>
> – the INVITE request
>
> – a 2xx or 18x response to an INVITE which contains a To tag,
>
> – the UPDATE request
>
> – a 2xx response to an UPDATE

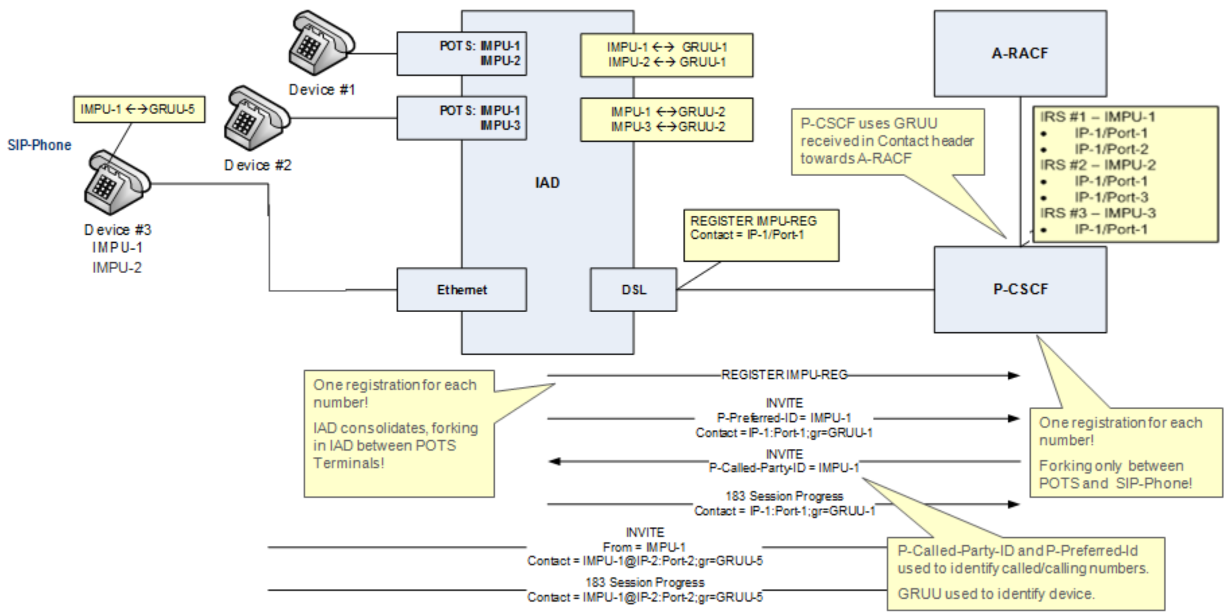

Figure 2 Principle of GRUU in SIP Messages

5. Support of network based Services

§4.2.2 Telephony

Voice over IP (VoIP) is performed in accordance with the SIP-Protocol. The specifications to be fulfilled for control of a communication are presented in section 7.

For the Media-Stream the Codecs G.711a [26] (A-Law) and G.722 [27] shall be used. For VGW (IAD) supporting ISDN accesses RFC 4040 [62] (Clearmode) shall be supported.

To avoid transcoding towards mobile users to AMR-WB Codec as described within 3GPP TS 26.090 [78] and TS 26.190 [80] should be implemented. 3GPP TS 26.171 [79] and TS 26.193 [81] will give further advice for implementation.

If AMR-WB is offered in an initial request G.722 shall be offered in addition within the initial SDP offer.

§4.2.3 Fax and Modem

For Fax and Modem transmission over IP, ITU-T Rec. V.152 [33] (based on G.711a [26]) shall be used. If the adjacent endpoint does not support ITU-T Rec. V.152 [33], Fax- and Modem connections shall be set up using G.711a (ITU-T Rec. T.30 [31]).

Currently, ITU-T Rec. T.38 [32] is not supported by the NGN platform of Deutsche Telekom. *Nevertheless T.38 shall be supported as possibility for sending Fax end-to-end as specified in [32], [76] and [77]. Important is that Release 3 of T.38 is implemented for interoperability reasons.*

§5 SIP Service functionality requirements

§5.1 General

The SIP service functionality requirements are defined in Annex B of the present document.

Further specific service requirements are described in the following.

~~*The Services Communication HOLD (Section C.2.6), Communicating Waiting (CW) (Section C.2.8), TOGGEL and CONF (Section C.2.2) shall be implemented as a End Client Service feature and as described within this specification.*~~

~~*The network centric feature logic for HOLD, CW, TOGGEL and CONF are not available, therefore these features must be additionally implemented locally on the SIP Client. This must be configured as default.*~~

~~*Based on the VGW (IAD) configuration it must be possible to activate a local/terminal based CW on a busy line if an INVITE without a CW indication is received by the VGW.*~~

§Annex C.2 Simulation services

~~*The Services Communication HOLD (Section C.2.6), Communicating Waiting (CW) (Section C.2.8), TOGGEL and CONF (Section C.2.2) shall be implemented as a End Client Service feature and as described within this specification.*~~

The network centric feature logic for HOLD, CW, TOGGEL and CONF is currently ~~*not*~~ available; therefore these features must be implemented ~~*locally on the SIP Client*~~ *as follows*.

§C.2.2 CONF (Conference)

The relevant standard for the CONF service is 3GPP TS 24.605 [16]. ~~*The current implementation does not support this Specification.*~~

~~*The 3PTY CONF MUST be supported by the VGW itself.*~~

~~*The context (focus) building of the RTP session of Party A, B and C has to apply within the VGW.*~~
NOTE: Concerning clause §4.5.2.1 "Actions at the originating UE", the relevant 3PTY procedures described in 1TR126 Annex B [3] shall apply in addition.

§C.2.8 CW (Communication Waiting)

Based on the VGW (IAD) configuration and it must be possible to activate a local/terminal based CW on a busy line if an INVITE without a CW indication is received by the VGW. The default for local/terminal based CW is *de*activated.

The relevant standard for the CW service is 3GPP TS 24.615 [22].

The sections "4.5.5.1 Actions at the UE of user C" and "4.5.5.3 Actions at the UE of user B" in 3GPP TS 24.615 [22] shall apply.

~~*The actions described within section "4.5.5.2 Actions at the AS of user B" of 3GPP TS 24.[22] are currently not executed within the IMS of DT. Nevertheless future network configuration could change, thus UE SHOULD be prepared to receive a Content-Type header field set to "application/vnd.3gpp.cw+xml" and apply the regarding procedures.*~~

~~*To avoid loss in QoS, a maximum of 2 active and 2 waiting communications shall apply.*~~

Note: To avoid "ghost ringing", an UE while in idle state should not signal an incoming communication (by any acoustical or visual signal, e.g. ringing signal or display information) based on a SIP request INVITE with XML: call-waiting-indication.

§Annex D.5 Call Waiting/Communication Waiting (CW)

§D.5.1 Procedures

~~*§D.5.1.0 General*~~

~~*All following described procedures for Activation, Deactivation, Interrogation and Invocation must be handled internally by the VGW/UE to process the Service. It shall not be sent towards the network. For a VGW it is recommended to implement those functions as follows.*~~

§D.5.1.1 Activation

(wait for dial tone) *43# (wait for ack.)

(wait for dial tone) *43*0# (wait for ack.) (for all VoIP lines)

§D.5.1.2. Deactivation

(wait for dial tone) #43# (wait for ack.)

(wait for dial tone) #43*0# (wait for ack.) (for all VoIP lines)

§D.5.1.3 Interrogation

(wait for dial tone) *#43# (wait for ack.)

§D.5.1.4 Invocation

§D.5.1.4.1 Acceptance of an incoming communication (with or without authorisation of 3PTY service)

(hang-up and wait for ringing signal)

§D.5.1.4.2 Acceptance of an incoming communication (with authorisation of 3PTY service)

a) (wait for special dial tone) **1** (the current communication will be released)

b) (wait for special dial tone) **2** (the current communication is put on HOLD)

§D.5.1.4.3 Rejection of an incoming communication

(wait for special dial tone) **0** (the incoming communication will be rejected)

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§Annex D.6 HOLD / TOGGLE

§D.6.1 Procedures

~~§D.6.1.0 General~~

~~All following described procedures for Activation, Deactivation, Interrogation and Invocation must be handled internally by the VGW/UE to process the Service. It shall not be sent towards the network. For a VGW it is recommended to implement those functions as follows.~~

§D.6.1.1 Invocation (…)

(wait for special dial tone and dial third party number) <DN> (0…9)

§D.6.1.1.1 Worst case (communication could not be established)

a) (wait for special dial tone) **1**

b) (hang-up and wait for ringing signal)

§D.6.1.2 Invocation (change to the party on HOLD –TOGGLE-)

(wait for special dial tone) **2** (the current communication is put on HOLD)

§D.6.1.3 Invocation (release a communication during HOLD)

§D.6.1.3.1 Invocation (release the current communication)

(wait for special dial tone) **1** (the communication on HOLD becomes active)

§D.6.1.3.2 Invocation (release the communication on HOLD)

(wait for special dial tone) **0** (the communication on HOLD will be released)

§D.6.1.3 Invocation (Release initiated by the current party)

Congestion tone provided

(wait for special dial tone) **1** or **2**

or

(wait for ringing tone)

§Annex D.7 Three Party Conference/Conference (3PTY/CONF)

§D.7.1 Procedures

~~*§D.7.1.0 General*~~

~~*All following described procedures for Activation, Deactivation, Interrogation and Invocation must be handled internally by the VGW/UE to process the Service. It shall not be sent towards the network. For a VGW it is recommended to implement those functions as follows.*~~

§D.7.1.1 Invocation (3PTY/CONF initiation)

Prerequisite: Initiator has one communication in an active state and a second communication on hold.

(wait for special dial tone) **3**

§D.7.1.2 Invocation (change from 3PTY/CONF to HOLD/TOGGLE)

(wait for special dial tone) **2**

Version	Published	Remarks
3.0.0		-locating P-CSCF and correct prioritization of P-CSCF in case of registration including maintenance procedures. -Preconditions support "passive" better described -Early-Media Header and indication of early media described to avoid misinterpretation. And allow handling of calls initiated by mobile devices. - use of from-change. No default setting - deletion of Annex A - Update of Annex B - Deletion of TS 124.503 - UPDATE to 3GPP Release 11 documents -Correction of *# Procedures using PIN (ECT, OCB, Kick Out, Black List, White List, ACR, CB, ICB) - CLIR 3 included in D.2.0 - Documentation Update TIP/TIR and OIP/OIR -MWI voided - Documentation Update of " 8.6 Support of NAT traversal by the UE" -MIME Type UPDATE Table 7-5 -UPDATE Table 7-4 SIP Headers - add references TR-069, TR-104 and TR-181 - add reference 3GPP TS 23.003 - C.2.8 allow implementations acting on "application/vnd.3gpp.cw+xml" All changes are backward compatible with the procedures described within 1TR114 Version 2.4.0
Amendment 1.2	07. January 2015	Additions to 1 TR 114 for the SIP REQUEST Retry Mechanism in Failure Cases
Amendment 2		pending
Amendment 3	09.02.2015	Implementation Guideline for use of preconditions and 100rel
Amendment 4	24. April 2015	Support of Bandwidth reservation, GRUU and centralized Services
Amendment 4.1	31. August 2016	Correction of URI Format for GRUU e.g. Contact < sip:contact-URI; gr=urn:uuid:**random-string>**