

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

1 TR 114

Version: 3.0.0

Amendment 5
(Reliable SIP & RTP Processing)

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1 Scope

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

2 Additions to 1 TR 114 for Reliable SIP Processing

The device shall support Reliable SIP request processing to be robust and Reliable against SIP attacks, e.g. SIP vicious.

The device shall only process SIP requests, received on its WAN interface, from P-CSCF source IPs (IPv4 or IPv6) which the Telekom VoIP account of the device is successfully registered to.

Note: The P-CSCF is the Deutsche Telekom IMS entity equivalent with a outbound proxy defined within SIP RFC 3261. The Deutsche Telekom used entity follows the 3GPP and ETSI TISPAN specifications. The needed UE procedures are defined within the main 1TR114 document.

In addition the device shall handle multiple registered conditions in edge scenarios, to process SIP requests from P-CSCF source

IPs (IPv4 or IPv6), e.g. due to synchronization time offset:

1. Telekom VoIP account X is registered to P-CSCF 1
 - 1.1 Telekom VoIP account Y is registered to P-CSCF 1
 - 1.2 Re-Register process for Telekom VoIP Account Y is succeeded to P-CSCF 1
 - 1.3 Re-Register process for Telekom VoIP account X is not possible to P-CSCF 1(P-CSCF 1 is temporarily unavailable), Telekom VoIP account X is then re-registering to P-CSCF 2 and also still registered to P-CSCF 1 until the valid register to P-CSCF 1 is expired.

In that case the VoIP account X is still registered to P-CSCF 1 and registered to P-CSCF 2

Additionally, all P-CSCF source IPs (IPv4 or IPv6) which are successfully resolved, based on device DNS resolution e.g. via SRV / AAAA / A records, are valid for P-CSCF source IP request processing. (under consideration of the TTL).

The device shall process SIP requests from SIP UA within the Home Network without any malfunction.

The device shall support Reliable SIP request processing with all valid SIP request methods acc. to RFC 3261, RFC 3262, RFC 6665, RFC 6086, RFC 3513, RFC 3311.

This P-CSCF origin based Reliable SIP request processing shall be valid for Telekom as VoIP provider.

2.2 UE receiving RTP from un-trusted sources

The UE device shall only process RTP & RTCP, received on its WAN interface, from A-BGF (Access Border Gateway Function) source IPs (IPv4 or IPv6) and Ports which were negotiated within the related SIP Session. RTP & RTCP Packets received from other sources shall be ignored and dropped.

NOTE : Depending on the network architecture an INVITE received may indicate different IP address for the sender of the SIP signalling and for the sender of the RTP & RTCP packets received.

Version	Published	Remarks
3.0.0		<ul style="list-style-type: none"> -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" <p>All changes are backward compatible with the procedures described within ITR114 Version 2.4.0</p>
Amendment 5	30.07.2014	Additions of Reliable SIP Processing