1 The scope of this document

The intention of this document is to provide a specification with a functional description of the interfaces at the passive Network Termination Points in NetCologne’s xDSL and fibre networks. It reflects the changes forced into law as of August 1st 2016. The document contains the description for the Access functionalities which are required to connect to the NetCologne network properly, as well as the required functionalities for Voice and Data Services. If there is any misleading information or a requirement for additional information or clarification, the reader of this document is kindly required to execute a formal request to NetCologne in order to be able to provide the required information. The figure below visualizes the scope of the descriptions.

2 Description of the Access

Line Port Description

In order to provide the various services to NetCologne’s customers by using copper and fiber access infrastructure, the following transmission standards and interfaces are used.

<table>
<thead>
<tr>
<th>Transmission type</th>
<th>Standards</th>
<th>Special characteristics</th>
</tr>
</thead>
<tbody>
<tr>
<td>ADSL 2+</td>
<td>G.992.5</td>
<td>• Annex B</td>
</tr>
<tr>
<td>G.SHDSL.bis</td>
<td>G.991.2</td>
<td>• IEEE 802.3 ah</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• TDMAM16(32)</td>
</tr>
<tr>
<td>VDSL2</td>
<td>G.993.2</td>
<td>• Depending on the type of network (FTTB / FTTC) different profiles are used: ADE17 and ADE30</td>
</tr>
<tr>
<td></td>
<td>G.993.5 (vector)</td>
<td>• In addition the PSD mask can be shaped individually between 2.2 and 17.6 MHz</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Transmission type</th>
<th>Standards</th>
<th>Special characteristics</th>
</tr>
</thead>
<tbody>
<tr>
<td>1000Base-X</td>
<td>IEEE 802.3z (LX/ZX/BX10 ... 80)</td>
<td>• LX, ZX, BX10 ... 80</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• IEEE 802.3ad (LAG)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• IEEE 802.1d</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• IEEE 802.1p/Q (tagging, QoS)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• IEEE 802.1w (nstp)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• G.8023Z (ring protection)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• Jumbo frames</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Transmission type</th>
<th>Standards</th>
<th>Special characteristics</th>
</tr>
</thead>
<tbody>
<tr>
<td>10GBase-X</td>
<td>IEEE 802.3ae (LR/ER/ZR)</td>
<td>• LR, ER, ZR</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• IEEE 802.3ad (LAG)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• IEEE 802.1d</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• IEEE 802.1p/Q (tagging, QoS)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• IEEE 802.1w (nstp)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• G.8023Z (ring protection)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• Jumbo frames</td>
</tr>
</tbody>
</table>

Table 1 Transmission standards for twisted pair

Table 2 Transmission standards for fibre

3 Description of Voice Services

This chapter describes the SIP interface between an User Equipment (UE) and the NGN Voice platform of NetCologne. It is based on 1TR114, regarding structure and content, but can be read without knowledge of 1TR114.

Chapter 4 incorporates a delta specification to TS 24.229. This specification can be used to implement a SIP-client on an IAD.

3.1 References

[3] DT 1TR126: Technical Specification for SIP User Equipments (UE) providing IMS simulation services via analogue (POTS) interfaces (POTS/SIP interworking) using the NGN platform of Deutsche Telekom
[13] 3GPP TS 24.229 V11.6.0 (2012-12) Annex B: Modified version for SIP (GM) interfaces provided by Deutsch Telekom only
## 3.2 Capabilities

### 3.2.1 SIP capabilities

The request of Preconditions (indication of SUPPORT/REQUIRED within an initial INVITE) are not part of this specification. Only passive support is required. Sending of preconditions supported or required with the initial INVITE SHALL NOT be done. Reliable Provisional Responses are mandatory. SIP URLs shall be supported in SIP header fields.

### 3.2.2 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 3.17.

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

### 3.2.3 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 NetCologne.

### 3.2.4 DTMF

For DTMF events RFC 4733 [65] and RFC 5244 [67] shall be supported. Note: In cases where the remote Endpoint does not support RFC4733 it shall be possible to send DTMF inband.

### 3.2.5 Early Media

For early media RFC 5009 MUST be supported. Due to the fact that not all functionalities shall support RFC5009 for early media further procedures for identifying early media need to be supported. In addition not in each case where an SDP is received within a provisional response early media apply.

Therefore the following procedures to identify if early media is received shall apply in the following sequence:

1. If a provisional response includes a P-Early-Media Header with “sendonly” and a require header with 100rel. The procedures shall apply with 3GPP TS 24,628 [17].
2. If a provisional response contains SDP and preconditions are not used.
3. Identifying if an RTP stream is received by the UE.

### 3.2.6 Locating of Proxy in case of (re-) Registration and change of P-CSF priority due to Maintenance

For proxy discovery and Registration Procedures the Specifications TS 24.229 [21], RFC3261 [71], RFC 2782 [47] and RFC 3263 [50] are valid.

The following procedures shall give a hint for End device vendors how to implement these procedures to fulfill the requirements of NetCologne.

Due to maintenance and failure situations the prioritization of P-CSF can change. Therefore the destination must be determined by applying the DNS procedures described within RFC3261 [71], RFC 2782 [47], RFC 3263 [50] and ANNEX B of this document.

A DNS query to request the actual SRV record set shall be done before sending a REGISTER or re-REGISTER request.

NOTE: This is valid in cases where the registration timer expires, or a network initiated deregistration was sent or in cases where final responses where received pointing to a failure situation where the target cannot be reached (e.g. 503 Response) or a redirect (305 response) was received.

As described within RFC 2782 [47] a client MUST attempt to contact the target host (P-CSF) with the lowest numbered priority it can reach; target hosts with the same priority SHOULD be tried in an order defined by the weight field.

Within NetCologne network normally only the priority field is used. TTL expiry shall be taken into consideration when starting Register and re-Register procedures.

### 3.2.7 Auto configuration

Out of scope

### 3.3 SIP Service functionality requirements

#### 3.3.1 General

The SIP service functionality requirements are defined in [21]. Further specific service requirements are described in the following.

CONF (Section C.2.2) shall be implemented as an End Client Service feature and as described within this specification.

The network centric feature logic for HOLD, CW, TOGEL 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 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.

#### 3.3.2 Direct Dial In (DDI)

Out of scope

#### 3.3.3 Codecs

Codecs listed in the following Table 3 are supported by the NGN platform of NetCologne.

NOTE: If no transcoding rules or other restrictions (e.g. RACS) contradict, any audio and video codecs will be transparently conveyed through the NGN platform of NetCologne.

<table>
<thead>
<tr>
<th>Specification</th>
<th>Title</th>
<th>Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711</td>
<td>Pulse code modulation (PCM) of voice frequencies</td>
<td>[26]</td>
</tr>
<tr>
<td>G.722</td>
<td>7 kHz Audio — Coding within 64 kBit/s</td>
<td>[27]</td>
</tr>
</tbody>
</table>

#### 3.4 Protocol (Profiles)

This section profiles the Gm interface for SIP UE intended to be connected to the NGN platform of NetCologne based on 3GPP TS 24.229 Release 11 [21] (endorsements).

Markings general used within the TEXT:

- Text modified due to NetCologne requirements that is added or deleted compared to 3GPP TS 24.229 Release 11 [21] is shown as cursive and underlined (example for added text).

For information: As usual within 3GPP Standards notes in Tables are mandatory and have to be implemented

#### 3.4.1 Modifications to 3GPP TS 24.229

The relevant modifications to 3GPP TS 24.229 [21] for SIP UE (Gm interface) intended to be connected to the NGN platform of NetCologne are provided in Annex B of the present document.

- procedures with the relevant Service Command Codes (SCC) (e.g. *214*). These service procedures (incl.SCC) are described in section 3.16 of this document. Either the SCC can be directly dialled via the key pad on the SIP Phone or via specific service menu buttons, which initiates the calling SCC. The SCC shall be sent in the format of a SIP URI: SCC@hostportion within an initial INVITE. If SIP equivalent procedures are available and supported by the network these shall be preferred.

- Particular services provided by the NGN platform of NetCologne require specific procedures using Switching Order Commands (SOC). A SIP UE supporting these services shall use the procedures (incl. SCC) which are described in section 3.16 of this document and 1TR126 [3].

- The Hook-Flash handling and the invocation of services are described in 1TR126 [3]. The implementation of the procedures for the Hook-Flash handling, if it is a real Hook-Flash or only a menu button to invoke the service; is a matter of the vendor and out of the scope of this document.

- Request URI = SIP URI with user=phone.
• For future network improvements the capabilities of registering and send- 
ing SIP URI as defined for Public User Identities in 3GPP TS 23.003 [75] 
SHOULD exist. Currently the only Format used is SIP URI’s representing a 
E.164 Number in the host portion. Default is SIP URI with user-phone 
• Header fields received my contain tel URI or alias URI as defined in 3GPP 
TS 23.003 [75] 
• All URI (Request, From etc.) should be presented within global number for- 
mate. 
• Request URI = SIP URI with user-phone is used for SCC like *(21)%23@ 
hostportion. 
• # (hash) in URIs must be escaped 
The Re-Ringing procedure for the SIP UE shall apply according to 1TR126 
[3]. 
The Protocol stacks shall work with IPv4 and IPv6. 
• Neither UIC or ISIM is applicable for this document. 
• iCSI and IARI are currently not used, but nevertheless elements included 
with SIP messages shall be passed on in compliance with the current 
specification. 
The Call-Id shall not include the own IP-address of the UA. 
• Network initiated De-Registration is part of this specification and must be 
supported. 
• Authentication shall be possible via HTTP Digest and without HTTP Digest 
(NASS bundled) based on the line/IP-Address. 
• Support of session timers regarding RFC 4028 [61] is mandatory. 
• For tones and announcements the procedures described in 3GPP TS 2.628, 
Annex D [17] shall apply. The bidirectional media shall be used. 
• Any final response either 200 OK or final error response (e.g. 4xx) shall 
close all existing early dialogs for the regarding Call-ID. 
• To avoid problems with a wide spread of existing clients 3PCC procedures 
shall only send INVITE without SDP and with 100rel supported so that the 
USAS can decide in sending reliable or unreliable provisional responses. 
• The AS shall send a 199 for the release of early dialogs. A further 18x re- 
sponse (e.g. 180 in case of CCNR activation rejection) may be sent after- 
wards. For UE 199 is mandatory to understand. 
The challenge mechanism shall be supported. 
• To avoid too many challenge cycles the nonce shall be included within each 
request during its validity. 
• DNS SRV capabilities (including TTL) shall be supported 
• HEX digits as defined within RFC 3966 [58] to be sent or received on the 
Gm interface in SIP URI user-phone are not allowed. 
• UE shall minimise or avoid REGISTER procedures for identifying fetch bind- 

g’s. The avoidance of this procedure is preferred to minimise the network 
load. 
• The restoration procedures as described in [21] shall be supported.

De-Registration
In cases where UE’s are booting, there is no knowledge if the UE is already 
registered or not. Therefore De-Register with *** in the contact header field is 
forbidden.

General procedure for REGISTER Message answered with a 403
General a 403 is an Indication that the user is not provisioned within the HSS. 
Nevertheless if 403 (Forbidden) has been received as a response to a REGIS-

TER request, a further registration attempts shall be done after 15 sec. In case 
further 403 responses received with the same URI in the Contact header field 
REGISTER requests are allowed with a random delay of 30- 60 minutes.

3.4.2 UE (Gm) interface, Profile tables based on 3GPP TS 24.229
In the following section the actual numbering of the endorsement document 
is kept with a leading “#” sign, if applicable. If not explicit noted, the refer-
ences mentioned within the following tables apply to [21 Annex B] i.e the 
numbering of reference [xyz] is the same as used within TS 24.229. T
The following status-codes shall be supported on the Gm-Interface:

<table>
<thead>
<tr>
<th>Item</th>
<th>PDU</th>
<th>Sending</th>
<th>Receiving</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Ref.</td>
<td>Profile status</td>
<td>UNI (Gm)</td>
</tr>
<tr>
<td>1</td>
<td>100 (Trying)</td>
<td>[26] 21.1.1</td>
<td>c21</td>
</tr>
<tr>
<td>2</td>
<td>180 (Ringing)</td>
<td>[26] 21.1.2</td>
<td>c2</td>
</tr>
<tr>
<td>3</td>
<td>181 (Call Is Being Forwarded)</td>
<td>[26] 21.1.3</td>
<td>c2</td>
</tr>
<tr>
<td>4</td>
<td>182 (Queued)</td>
<td>[26] 21.1.4</td>
<td>c2</td>
</tr>
<tr>
<td>5</td>
<td>183 (Session Progress)</td>
<td>[26] 21.1.5</td>
<td>c1</td>
</tr>
<tr>
<td>5A</td>
<td>199 (Early Dialog Terminated)</td>
<td>[142] 8</td>
<td>c32</td>
</tr>
<tr>
<td>6</td>
<td>200 (OK)</td>
<td>[26] 21.2.1</td>
<td>m</td>
</tr>
</tbody>
</table>

Table 4 Supported Methods

Conditions for Table 4.2.1:

c1: IF received reject with 405.
c2: IF received then response shall be ignored.
c3: IF received reject with 501.
c4: Void.
c5: IF NetCologne applications using INFO THEN m ELSE c1.
c6: IF NetCologne applications using NOTIFY THEN m ELSE n/a.
c7: IF Preconditions or Tones and Announcements (early media) with 18x THEN m ELSE o.
c8: IF NetCologne applications using NOTIFY THEN m ELSE n/a.
c9: IF NetCologne applications using NOTIFY THEN m ELSE c1.
c10: needed for future services

c11: needed for future services

c12: IF ECT or other application using REFER message THEN m ELSE c1.
c13: IF ECT or other application using REFER message THEN m ELSE n/a.
c14: IF Presence or other application using Publish THEN m ELSE c1.
c15: IF Presence or other application using Publish THEN m ELSE n/a.
c16: IF [22] TABLE A.4/1 in [21 Annex B] THEN m ELSE n/a -- client behaviour for registration.
c17: IF [22] TABLE A.4/17 in [21 Annex B] THEN m ELSE n/a -- the SIP UPDATE method?

NOTE 1: The OPTION method is used to check reliability between UE and PCSCF.
Table 5 Supported status-codes

Conditions for Table 4.2-2:

c1: IF TABLE 4.2-1/9 THEN m ELSE n/a - INVITE response.
c2: IF TABLE 4.2-1/9 THEN o ELSE n/a - INVITE response.
c3: IF [22] TABLE A.4 /20 THEN m ELSE n/a - SIP specific event notification extension.
c4: IF TABLE 4.2-1/19 OR TABLE 4.2-1/21 THEN m ELSE n/a - REGISTER response or SUBSCRIBE response.
c5: IF [22] TABLE A.4 /37 AND [22] TABLE A.4 /2 THEN m ELSE n/a - security mechanism agreement for the session initiation protocol and registrar.
c6: IF [22] TABLE A.4 /37 THEN m ELSE n/a - security mechanism agreement for the session initiation protocol.
c7: IF [22] TABLE A.4 /42 AND (TABLE 4.2-1/9 OR TABLE 4.2-1/23) THEN m ELSE n/a - the SIP session timer AND INVITE response OR UPDATE response.
c8: IF [22] TABLE A.4 /43 AND TABLE 4.2-1/17 THEN m ELSE n/a - the SIP Referred-By mechanism and REFER response.
c9: IF [22] TABLE A.4 /43 AND TABLE 4.2-1/17 THEN m ELSE n/a - the SIP Referred-By mechanism and REFER response.
c10: IF [22] TABLE A.4 /44 THEN m ELSE n/a - the Session Initiation Protocol (SIP) "Replaces" header.
c11: IF TABLE 4.2-1/9 THEN m ELSE n/a - INVITE response.
c12: IF [22] TABLE A.3 /4 THEN m ELSE n/a - S-CSCF.
c13: IF [22] TABLE A.3 /4 THEN m ELSE n/a - rejecting anonymous requests in the session initiation protocol.
c20: IF [22] TABLE A.4 /41 THEN m ELSE n/a - an event state publication extension to the session initiation protocol.
c21: IF TABLE 4.2-1/9 OR TABLE 4.2-1/9B OR TABLE 4.2-1/13 OR TABLE 4.2-1/15B OR TABLE 4.2-1/17 OR TABLE 4.2-1/19 OR TABLE 4.2-1/21 THEN m ELSE n/a - INVITE response or MESSAGE response or OPTIONS response or PUBLISH response or REFER response or REGISTER response or SUBSCRIBE response.
c22: IF [22] TABLE A.4 /57 THEN m ELSE n/a - managing client initiated connections in SIP.
c23: IF [22] TABLE A.4 /60 THEN m ELSE n/a - SIP location conveyance.
c24: IF CDV THEN m ELSE n/a - 

c25: UE may use it if internal forwarding applies.
c26: IF [22] TABLE A.4 /75B THEN m ELSE n/a - a recipient within the framework for consent-based communications in SIP.
c27: IF [22] Table A.4 /75A THEN m ELSE n/a - a relay within the framework for consent-based communications in SIP.
c28: IF [22] TABLE A.4 /2 AND [22] TABLE A.4 /57 THEN m ELSE n/a - registrar, managing client initiated connections in SIP.
c29: IF [22] TABLE A.4 /1 AND [22] TABLE A.4 /57 THEN m ELSE n/a - client behaviour for registration, managing client initiated connections in SIP.
c31: IF [22] TABLE A.4 /71 THEN m ELSE n/a - addressing an amplification vulnerability in session initiation protocol forking proxies.
c32: IF [22] TABLE 4.2-1/9 AND [22] TABLE A.4 /81 THEN m ELSE n/a - INVITE response and 199 (Registration Complete) response.
c33: IF [22] TABLE A.4 /13 THEN m ELSE n/a - SIP INFO method and package framework.
c34: IF [22] TABLE A.4 /16 OR [22] TABLE A.3 /6 THEN m ELSE IF [22] TABLE 4.2-1/9 THEN m ELSE n/a - initiating a session which require local and/or remote resource reservation, MGCF, INVITE response.
c35: IF [22] TABLE A.4 /16 THEN m ELSE n/a - integration of resource management and SIP.
3.4.2.4 Support of SIP Headers on the UNI (GM) – Interface

<table>
<thead>
<tr>
<th>Item</th>
<th>Header</th>
<th>Sending (UE to P-CSCF)</th>
<th>Receiving (P-CSCF to UE)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Ref.</td>
<td>Profile status</td>
<td>Ref.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>UNI (GM)</td>
<td></td>
</tr>
<tr>
<td>1</td>
<td>Accept</td>
<td>[26] 20.1</td>
<td>m m</td>
</tr>
<tr>
<td>2</td>
<td>Accept-Contact</td>
<td>[56B] 9.2</td>
<td>o o</td>
</tr>
<tr>
<td>3</td>
<td>Accept-Encoding</td>
<td>[26] 20.2</td>
<td>o o</td>
</tr>
<tr>
<td>4</td>
<td>Accept-Language</td>
<td>[26] 20.3</td>
<td>o o</td>
</tr>
<tr>
<td>5</td>
<td>Alert-Info</td>
<td>[26] 20.4</td>
<td>o m</td>
</tr>
<tr>
<td>6</td>
<td>Allow</td>
<td>[26] 20.5, [26] 5.1</td>
<td>m m</td>
</tr>
<tr>
<td>7</td>
<td>Allow-Events</td>
<td>[26] 7.2.2</td>
<td>o m</td>
</tr>
<tr>
<td>7b</td>
<td>Answer-Mode</td>
<td>[158] o o</td>
<td>[158] o o</td>
</tr>
<tr>
<td>8</td>
<td>Authentication-In-Info</td>
<td>[26] 20.6</td>
<td>o m</td>
</tr>
<tr>
<td>9</td>
<td>Authorization</td>
<td>[26] 20.7</td>
<td>m m</td>
</tr>
<tr>
<td>10</td>
<td>Call-ID</td>
<td>[26] 20.8</td>
<td>m m</td>
</tr>
<tr>
<td>11</td>
<td>Call-Info</td>
<td>[26] 20.9</td>
<td>o n/a</td>
</tr>
<tr>
<td>12</td>
<td>Contact</td>
<td>[26] 20.10</td>
<td>m m</td>
</tr>
<tr>
<td>13</td>
<td>Content-Disposition</td>
<td>[26] 20.11</td>
<td>m m</td>
</tr>
<tr>
<td>14</td>
<td>Content-Encoding</td>
<td>[26] 20.12</td>
<td>o m</td>
</tr>
<tr>
<td>15</td>
<td>Content-Language</td>
<td>[26] 20.12</td>
<td>o m</td>
</tr>
</tbody>
</table>

16 Content-Length [26] 20.14 m m [26] 20.14 m m
17 Content-Type [26] 20.15 m m [26] 20.15 m m
18 Cseq [26] 20.16 m m [26] 20.16 m m
19 Date [26] 20.17 o m [26] 20.17 o m
20 Error-Info [26] 20.18 o o [26] 20.18 o m
21 Event [26] 8.2.1 o m [26] 8.2.1 o m
22 Expires [26] 20.19 o m [26] 20.19 o m
23 From [26] 20.20 m m [26] 20.20 m m
23A Geolocation [89] 3.2 n/a n/a [89] 3.2 n/a n/a
23B Geolocation-Routing [89] 4.2 n/a n/a [89] 4.2 n/a n/a
24 History-Info [66] 4.1 n/a n/a [66] 4.1 o m
25 In-Reply-To [26] 20.21 o o [26] 20.21 o o
26 Join [61] 7.1.1 o o [61] 7.1.1 o o
26b Max-Breadth [117] o o [117] o n/a n/a
27 Max-Forwards [26] 20.22 m m [26] 20.22 m m
28 MIME-Version [26] 20.24 o m [26] 20.24 o m
29 Min-Expires [26] 20.25 [70] 5.6 m m [70] 5.6 m m
30 Min-SE [58] 5 o o [58] 5 m m
31 Organization [26] 20.25 o o [26] 20.25 o o
32 P-Access-Network-Info [52] 4.4 o o [52] 4.4 o m
32a P-Answer-State [34] 9.1 n/a n/a [34] 9.1 n/a n/a
33 P-Asserted-Identity [34] 9.1 n/a n/a [34] 9.1 o m
33a P-Asserted-Service [121] n/a n/a [121] n/a c1
33b P-Associated-URI [52] 4.1 n/a n/a [52] 4.1 c9 m
34 P-Called-Party-ID [52] 4.2 n/a n/a [52] 4.2 n/a c1
35 P-Charging-Func-Addresses [52] 4.5 n/a n/a [52] 4.5 n/a c1
36 P-Charging-Vector [52] 4.6 n/a n/a [52] 4.6 n/a c1
36b P-Early-Media [109] 8 o m [109] 8 o m
38 P-Media-Authorization [31] 6.1 o o [31] 6.1 o o
39 P-Preferred-Identity [34] 9.2 m m [34] 9.2 n/a n/a
39a P-Preferred-Service [121] 4.2 n/a n/a [121] 4.2 n/a c1
39b P-Profile-Key [97] 5 n/a n/a [97] 5 n/a c1
39c P-User-Database [82] 4 n/a n/a [82] 4 n/a c1
40 P-Visited-Network-ID [52] 4.3 n/a n/a [52] 4.3 n/a c1
40a Path [35] 4.2 o o [35] 4.2 m m
41 Priority [26] 20.26 n/a n/a [26] 20.26 n/a c1
41a Priv-Answer-Mode [158] o o [158] o o
Table 6 Supported Headers

<table>
<thead>
<tr>
<th>Item</th>
<th>Ref.</th>
<th>Profile status UNI (GM)</th>
<th>Sending (UE to P-CSCF)</th>
<th>Receiving (P-CSCF to UE)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Note 1</td>
<td>c1</td>
<td>o</td>
<td>Note 1</td>
</tr>
<tr>
<td>4</td>
<td>1TR126 [3]</td>
<td>m</td>
<td>o</td>
<td>1TR126 [3]</td>
</tr>
<tr>
<td>6</td>
<td>RFC 2327 [43]</td>
<td>m</td>
<td>m</td>
<td>RFC 2327 [43]</td>
</tr>
<tr>
<td>7</td>
<td>RFC 3863 [57]</td>
<td>n/a</td>
<td>n/a</td>
<td>RFC 3863 [57]</td>
</tr>
<tr>
<td>8</td>
<td>multipart/mixed</td>
<td>m</td>
<td>m</td>
<td>m</td>
</tr>
<tr>
<td>9</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>10</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>11</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>12</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Table 3.2.5 MIME Types

The following MIME Types shall be supported:

- application/vnd.etsi.simservs+xml
- application/vnd.etsi.aoc+xml
- application/vnd.etsi.pstn+xml
- application/vnd.etsi.watcherm.info+xml
- application/watcherm.info+xml
- image/t.38
- image/x-watcherm.info+xml
- image/vnd.etsi.cdw+xml
- image/vnd.etsi.cdm+xml
- image/vnd.etsi.ccm+xml

Note 1: The use is only foreseen for NetCologne domain.

Note 2: void.

Note 3: P-Answer-State header extension to the session initiation protocol for the open mobile alliance push to talk over cellular.

Note 4: void.

Note 5: This Reference is shown within Section 2 of this document, because this is a requirement of NetCologne to align Calls all over the network.
3.4.2.6 SDP Types

<table>
<thead>
<tr>
<th>Item</th>
<th>Type</th>
<th>Sending</th>
<th>Receiving</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Ref. RFC</td>
<td>Status</td>
</tr>
<tr>
<td></td>
<td></td>
<td>status</td>
<td></td>
</tr>
</tbody>
</table>

### Session level description

| 1 | v= (protocol version) | (39) 5.1 | m | m | (39) 5.1 | m | m |
| 2 | o= (owner/creator and session identifier) | (39) 5.2 | m | m | (39) 5.2 | m | m |
| 3 | c= (session name) | (39) 5.3 | m | m | (39) 5.3 | m | m |
| 4 | i= (session information) | (39) 5.4 | o | c2 | (39) 5.4 | m | c3 |
| 5 | u= (URI of description) | (39) 5.5 | o | c4 | (39) 5.5 | o | n/a |
| 6 | e= (email address) | (39) 5.6 | o | c4 | (39) 5.6 | o | n/a |
| 7 | p= (phone number) | (39) 5.6 | o | c4 | (39) 5.6 | o | n/a |
| 8 | c= (connection information) | (39) 5.7 | c5 | c5 | (39) 5.7 | m | m |
| 9 | b= (bandwidth information) | (39) 5.8 | o | o | (NOTE 1) | (39) 5.8 | m | m |

### Time description (one or more per description)

| 10 | t= (time the session is active) | (39) 5.9 | m | m | (39) 5.9 | m | m |
| 11 | r= (zero or more repeat times) | (39) 5.10 | o | c4 | (39) 5.10 | o | n/a |

### Session level description (continued)

| 12 | z= (time zone adjustments) | (39) 5.11 | o | n/a | (39) 5.11 | o | n/a |
| 13 | k= (encryption key) | (39) 5.12 | x | x | (39) 5.12 | n/a | n/a |
| 14 | a= (zero or more session attribute lines) | (39) 5.13 | o | o | (39) 5.13 | m | m |

### Media description (zero or more per description)

| 15 | m= (media name and transport address) | (39) 5.14 | o | o | (39) 5.14 | m | m |
| 16 | n= (media title) | (39) 5.4 | o | c2 | (39) 5.4 | o | c3 |
| 17 | c= (connection information) | (39) 5.7 | c1 | c1 | (39) 5.7 | c1 | c1 |
| 18 | b= (bandwidth information) | (39) 5.8 | o | (NOTE 1) | (39) 5.8 |
| 19 | k= (encryption key) | (39) 5.12 | x | x | (39) 5.12 | n/a | n/a |

### Table 8 SDP Types

<table>
<thead>
<tr>
<th>Item</th>
<th>Type</th>
<th>Sending</th>
<th>Receiving</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Ref. RFC</td>
<td>Status</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Ref. RFC</td>
<td>Status</td>
</tr>
</tbody>
</table>

- a= (zero or more media attribute lines) | [39] 5.13 | o | o | [39] 5.13 | m | m |

**Conditions for Table 4.2-5**

- **c1**: If Table 4.2-5/15 AND NOT Table 4.2-5/8 THEN m ELSE m
- **c2**: IF [22] Table A.3A/6 THEN c ELSE o
- **c3**: IF [22] Table A.3A/6 THEN n/a ELSE m
- **c4**: IF [22] Table A.3A/6 THEN x ELSE o

**Note 1**: 3GPP the definition is within 3GPP TS 29.163 [24].

**Note 2**: Other MIME Types can be received and must be discarded in case where no content disposition header is present the MIME is not known.
NOTE 1: Further specification of the usage of this attribute is defined by specifications relating to individual codecs.

### 3.4.3 SIP User Agent (UA)

#### 3.4.3.1 Supported SIP Signalling Transport Protocols in UA

The following SIP Signalling Transport Protocols shall be supported:

<table>
<thead>
<tr>
<th>Protocol (NOTE)</th>
<th>Specification</th>
<th>Ref.</th>
<th>Support</th>
</tr>
</thead>
<tbody>
<tr>
<td>UDP</td>
<td>RFC 0768/STD006</td>
<td>[34]</td>
<td>m</td>
</tr>
<tr>
<td>TCP</td>
<td>RFC 0763/STD007</td>
<td>[37]</td>
<td>m</td>
</tr>
<tr>
<td>TLS</td>
<td>RFC 2246</td>
<td>[42]</td>
<td>o</td>
</tr>
<tr>
<td>SCTP</td>
<td>ETSI TS 102 144</td>
<td>[5]</td>
<td>o</td>
</tr>
<tr>
<td>IPsec</td>
<td>RFC 2411</td>
<td>[44]</td>
<td>o</td>
</tr>
</tbody>
</table>

Note: The following combinations shall be possible to configure:
- SIP over UDP
- SIP over TCP without TLS
- SIP over TCP with TLS

Table 10 Supported Signalling Transport Protocols in UA

#### 3.4.3.2 Support of IPv4 and IPv6

Currently IPv6 for Voice is not supported.

<table>
<thead>
<tr>
<th>Specifi- cation</th>
<th>Title</th>
<th>Ref.</th>
<th>Support</th>
</tr>
</thead>
<tbody>
<tr>
<td>RFC 0791</td>
<td>Internet Protocol, Version 4</td>
<td>[35]</td>
<td>m</td>
</tr>
<tr>
<td>RFC 0792</td>
<td>Internet Control Message Protocol</td>
<td>[36]</td>
<td>m</td>
</tr>
<tr>
<td>RFC 1035</td>
<td>Domain name implementation and specification</td>
<td>[38]</td>
<td>m</td>
</tr>
<tr>
<td>RFC 2460</td>
<td>Internet Protocol, Version 6</td>
<td>[46]</td>
<td>n/a</td>
</tr>
<tr>
<td>RFC 2782</td>
<td>A DNS RR for specifying the location of services (DNS SRR)</td>
<td>[47]</td>
<td>m</td>
</tr>
<tr>
<td>RFC 2915</td>
<td>The Naming Authority Pointer (NAPTR) DNS Resource Record</td>
<td>[48]</td>
<td>o</td>
</tr>
<tr>
<td>RFC 3596</td>
<td>DNS Extensions to Support IP Version 6</td>
<td>[54]</td>
<td>n/a</td>
</tr>
<tr>
<td>RFC 4443</td>
<td>Internet Control Message Protocol (ICMPv6) for the Internet Protocol Version 6 (IPv6) Specification; March 2006</td>
<td>[64]</td>
<td>n/a</td>
</tr>
<tr>
<td>RFC 4884</td>
<td>Extended ICMP to Support Multi-Part Messages, April 2007</td>
<td>[70]</td>
<td>m</td>
</tr>
</tbody>
</table>

Table 11 RFC for support of IPv4 and IPv6
Technical Specification of the interfaces between User Equipment and the NGN Platforms of NetCologne / V0.3 / aicabg / Änderungen vorbehalten

Technical Specification of the interfaces between User Equipment and the NGN Platforms of NetCologne / V0.3 / aicabg / Änderungen vorbehalten

This chapter describes the service functionality requirements as a recommendation for the behavior of SIP user equipments connected to NGN Voice platform of NetCologne.

3.4.3.3 Video Codec Transport Procedures

Video is not supported yet.

3.4.3.4 Real Time Transport Procedures

<table>
<thead>
<tr>
<th>Specification</th>
<th>Title</th>
<th>Ref.</th>
<th>Support</th>
</tr>
</thead>
<tbody>
<tr>
<td>RFC 4040</td>
<td>RTP Payload Format for a 64kbit/s Transparent Call; April 2005 (see Note)</td>
<td>[62]</td>
<td>c1</td>
</tr>
</tbody>
</table>

Conditions:
- c1: If ISDN interworking then m else o.

NOTE1: This protocol is applicable to carry 64 kbit/s channel data transparently in RTP packets, using a pseudo-codec called “Clearmode” and is used in case of ISDN accesses via IADs, only.

NOTE2: Fragmented IP packets are not supported by the NGN platform of NetCologne. If the UA chooses to send RTCP/SDP packets it shall not send the UAs public IP address.

Table 14 Specification Real-time Transport Procedures

3.5 IAD- requirements

SIP terminal can only be connected to NetCologne NGN via IAD’s (see scope of document). IAD communicates with NC NS via a private IP address (DHCP) in the Voice network. Private IP addresses are IPv4 addresses.

If the IAD has to register several subscribers each of them is identified via its own contact. Fragmentation should not be used.

3.5.1 Network access

In general Quality of Service is assured by using different VLAN’s for Voice and data prioritization. To guarantee best service packetization size should be 20ms and codec G711a must be used.

The IAD is responsible for Prioritisation & marking of traffic:
- Voice Control Class 6 (DSCP 110 000)
- Voice bearer Class 5 (DSCP 101 110)
- PPP/PPPoE Control Traffic Class 6
- Best Effort Class 0 (DSCP 000 000)

3.5.2 Number Handling by the UE

Numbers should be sent “as dialed” by the user. No modification is necessary!

3.5.3 Support of NAT traversal

is not supported (and not necessary in the environment in scope)

3.6 Interworking requirements for SIP user equipment (UE)

The interworking requirements for SIP user equipment (e.g. IAD) are specified in separate documents. The referenced documents can be interpreted as recommendations for SIP terminal developers and vendors.

3.6.1 Analogue (POTS – SIP basic interworking requirements

The Analogue (POTS) – SIP basic interworking requirements are contained in the technical specification 1 TR 126 [3].

3.6.2 DSS1 – SIP basic interworking requirements

The DSS1 – SIP basic interworking requirements are contained in the technical specification 1 TR 127 [4].

3.7 Service functionality requirements

This chapter describes the service functionality requirements as a recommendation for the behavior of SIP user equipments connected to NGN Voice platform of NetCologne.

The relevant service code commands (SCC) for provision/withdrawal, registration/erasure, activation/de-activation, interrogation and invocation are provided in 3.12. Most of the services can also be administered via Web interface.

3.7.1 Calling Line Identification Restriction / Originating Identification Restriction (CLIR/OIR)

3.7.1.1 Description

CLIR can be implemented in several flavors. NetCologne implemented the following:

CLIR 2: While calling line identification presentation is set to permanent mode, the subscriber can activate the calling line identification restriction on per communication basis. If this service is activated for an outgoing communication, the calling line identification is restricted for this communication; after this communication the CLIR is automatically deactivated, again.

CLIR 3: Permanent calling line identification restriction: This service does NOT allow the subscriber to activate or deactivate the CLIR service permanently.

3.7.1.2 Procedures

3.7.1.3 Activation

3.7.1.3.1 CLIR2/OIR2

(wait for dial tone) *31*<DN>

3.7.1.4 Deactivation

3.7.1.4.1 CLIR2/OIR2

Automatically

3.7.1.5 Interrogation

Not applicable

3.7.2 Connected Line Identification Presentation / Terminating Indication Presentation (COLP/TIP)

Not supported

3.7.3 Connected Line Identification Restriction / Terminating Indication Restriction (CLIR/TIR)

Not supported

3.7.4 Call Waiting /Communication Waiting (CW)

All of the following procedures for Activation, Deactivation, Interrogation and Invocation must be handled as described internally by the VGW/UE to process the service. The service codes should not be sent to the network. For an IAD it is recommended to implement those functions as follows.

3.7.4.1 Activation

- <pick up> (wait for dial tone) *43* (wait for ack.) <hang up>
- <pick up> (wait for dial tone) *43*<DN> (wait for ack.) <hang up> (for all VoIP lines)

3.7.4.2 Deactivation

- <pick up> (wait for dial tone) #43# (wait for ack.) <hang up>
- <pick up> (wait for dial tone) #43*# (wait for ack.) <hang up> (for all VoIP lines)

3.7.4.3 Interrogation

- <pick up> (wait for dial tone) *43# (wait for ack.) <hang up>
- <pick up> (wait for dial tone) #43# (wait for ack.) <hang up>

3.7.4.4 Invocation

3.7.4.5 Acceptance of an incoming communication (with or without authorisation of 3pTY service)

- <hang up> (hang-up and wait for ringing signal) <pick up>

3.7.4.6 Acceptance of an incoming communication (with authorisation of 3pTY service)

a) <hook-flash> (wait for special dial tone) 1 (the current communication will be released)
b) <hook-flash> (wait for special dial tone) 2 (the current communication is put on HOLD)

3.7.4.7 Rejection of an incoming communication
<hook-flash> (wait for special dial tone) 0 (the incoming communication will be rejected)

3.7.5 Hold / Toggle
All following described procedures for Activation, Deactivation, Interrogation and Invocation must be handled internally by the IAD/UE to process the service. The service codes should not be sent to the network. For a IAD it is recommended to implement these functions as follows.

3.7.5.1 Invocation (….)
<hook-flash> (wait for special dial tone and dial third party number) <DN> (0…9)

3.7.5.2 Worst case (communication could not be established)
a) <hook-flash> (wait for special dial tone) 1
b) <hang up> (hang-up and wait for ringing signal) <pick up>

3.7.5.3 Invocation (change to the party on HOLD – TOGGLE)
<hook-flash> (wait for special dial tone) 2 (the current communication is put on HOLD)

3.7.5.4 Invocation (release a communication during HOLD)
<hook-flash> (wait for special dial tone) 1 (the communication on HOLD becomes active)

3.7.5.5 Invocation (release the communication on HOLD)
<hook-flash> (wait for special dial tone) 0 (the communication on HOLD will be released)

3.7.5.6 Invocation (Release initiated by the current party)
Congestion tone provided
<hook-flash> (wait for special dial tone) 1 or 2
or
<pick up> (wait for ringing tone) <hang up>

3.7.6 Three Party Conference/Conference (3PTY/CONF)
All of the following procedures for Activation, Deactivation, Interrogation and Invocation must be handled as described internally by the IAD/UE to process the service. The service codes should not be sent to the network. For an IAD it is recommended to implement these functions as follows.

3.7.6.1 Invocation (3PTY/CONF initiation)
Prerequisite: Initiator has one communication in an active state and as second communication on hold.
<hook-flash> (wait for special dial tone) 3

3.7.6.2 Invocation (change from 3PTY/CONF to HOLD/TOGGLE)
<hook-flash> (wait for special dial tone) 2

3.7.7 Communication Diversion: Call Forwarding Unconditional / Communication Forwarding Unconditional (CDIV:CFU)

3.7.7.1 Activation
<pick up> (wait for dial tone) *61*<CFN># (wait for ack.) <hang up>

3.7.7.2 Deactivation
<pick up> (wait for dial tone) #61# (wait for ack.) <hang up>

3.7.7.3 Interrogation
<pick up> (wait for dial tone) *#61# (wait for ack.) <hang up>

3.7.8 Communication Diversion: Call Forwarding Busy / Communication Forwarding Busy (CDIV:CFB)

3.7.8.1 Activation
<pick up> (wait for dial tone) *67*<CFN># (wait for ack.) <hang up>

3.7.8.2 Deactivation
<pick up> (wait for dial tone) #67# (wait for ack.) <hang up>

3.7.8.3 Interrogation
<pick up> (wait for dial tone) *#67# (wait for ack.) <hang up>

3.7.9 Communication Diversion: Call Forwarding No Reply / Communication Forwarding No Reply (CDIV:CFNR)
The default timer for CFNR is 20 seconds and can be changed via Web Interface.

3.7.9.1 Activation
<pick up> (wait for dial tone) *61*<CFN># (wait for ack.) <hang up>

3.7.9.2 Deactivation
<pick up> (wait for dial tone) #61# (wait for ack.) <hang up>

3.7.9.3 Interrogation
<pick up> (wait for dial tone) *#61# (wait for ack.) <hang up>

3.7.10 Malicious Communication Identification

3.7.10.1 Activation
<pick up> (wait for dial tone) 08# <hang up>

3.7.10.2 Deactivation
Not applicable

3.7.10.3 Interrogation
Not applicable