NICC ND 1442 V1.1.1 (2024-01)

NICC Document

SIP Implementers Guide

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Contents

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Foreword

This NICC Document (ND) has been produced by NICC SIP Task Group.

Introduction

Communication Providers (CPs) and customers who utilise SIP trunks often refer to RFCs as a guideline for implementing SIP/SDP. However, they may encounter ambiguity and interoperability issues when it comes to SIP/SDP message exchanges and session characterisation that they wish to address. It is crucial to ensure that the requested action for a session is well understood and correctly executed.

The present document serves as a guidance for UK CPs to establish a common ground and implementation approach for certain SIP message exchanges that may create or modify sessions, thereby achieving the desired actions or expected outcomes. It is important to note that this document does not replace the SIP/SDP message methods outlined in RFCs or published NICC Documents (NDs). Instead, it serves as a reference for those seeking advice on implementation and compliance with standards. It is intended to be a guide that parties can adopt and agree upon when faced with known ambiguity and interoperability issues.

1 Scope

The present document gives guidance to UK Communication Providers (CPs) on how to use SIP (Session Initiation Protocol) on UK User Network Interface (UNI) and Network to Network Interface (NNI) connections and appropriate inter-workings.

This is a guidance document, thus if there is any discrepancy with any NICC normative specifications, the NICC normative specification should be treated as authoritative.

2 References

2.1 Normative references

The following referenced documents are indispensable for the application of this document. For dated references, only the edition cited applies. For non-specific references, the latest edition of the referenced document (including any amendments) applies.

[1] ND1035 SIP Network to Network Interface Signalling
[2] ND1033 NGA Telephony SIP User Profile
[3] ND1034 UK SIPconnect Endorsement
[4] RFC 3264 An Offer/Answer Model with the Session Description Protocol (SDP)
[5] RFC 3261 SIP: Session Initiation Protocol.
[6] ITU-T Rec Q.118 Abnormal conditions – Special release arrangements
[7] ND1657 SIP – Overload Control

2.2 Informative references

3 Definitions, symbols and abbreviations

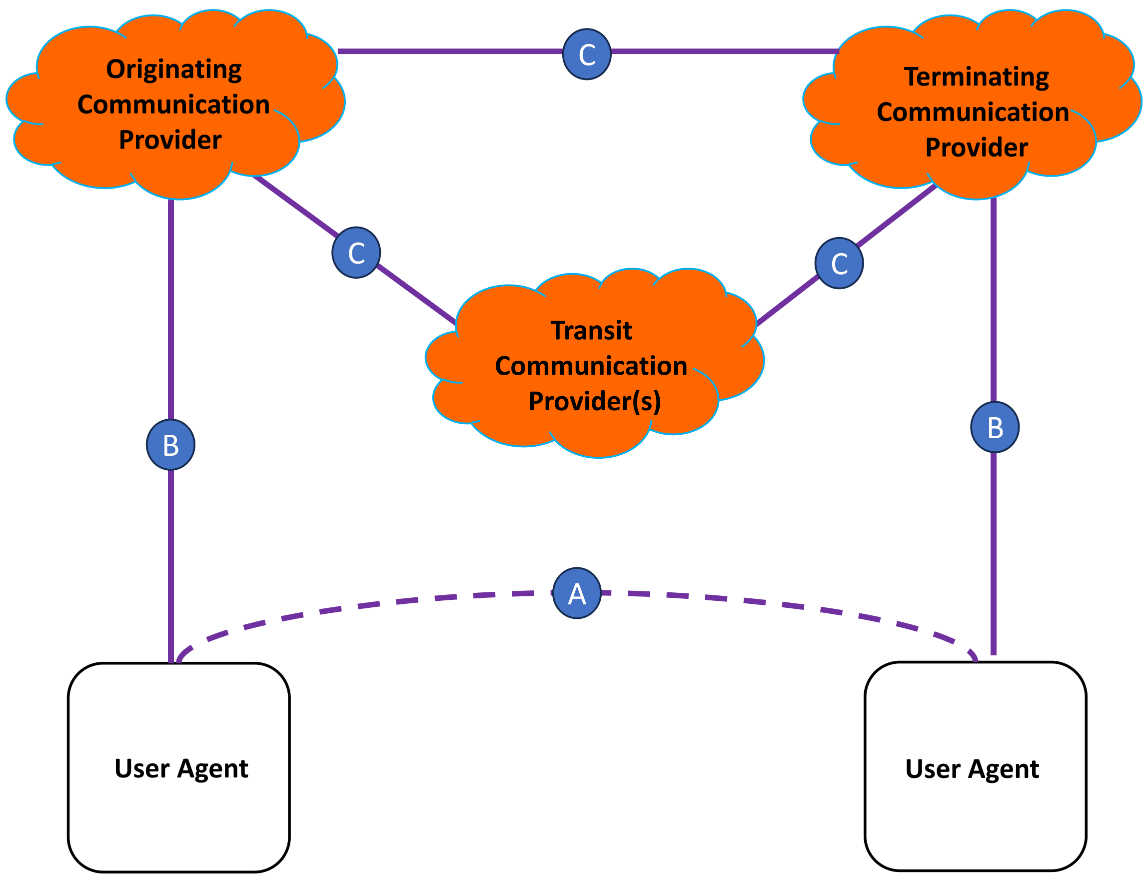
3.1 Abbreviations

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

CP	Communication Providers
NNI	Network to Network Interface
SBC	Session Border Controller
SDP	Session Description Protocol
SIP	Session Initiation Protocol
UA	User Agent
UAS	User Agent Server
UNI	User to Network Interface

4 Network Architecture

The following diagram provides a high-level SIP network architecture, whose purpose is intended to provide reference point(s) which can then be used to indicate the applicability of use cases that are covered in this document



The following reference points are identified in the above network architecture diagram:

- Reference Point A
  - This relates to the logical, end-to-end connection that exists between two user agents involved in a SIP call
- Reference Point B
  - This relates to the User-Network Interface (UNI) that exists between a user agent and its associated Communication Provider (CP). The same reference point applies regardless of whether the CP in question is associated with the originating or the terminating user agent for a given SIP call
- Reference Point C
  - This relates to the Network-Network Interface (NNI) interface that exists between Communication Providers. The same reference point applies regardless of whether the CPs in question are either the originating CP, a transit CP or the terminating CP

5 Use Cases

5.1 Should the called number be sent using enbloc or overlap?

Network Architecture Reference Points: A

The SIP protocol allows the called number to be sent as either enbloc or overlap;

- enbloc means that the whole number is sent in one message
- overlap means that parts of the number are sent, in sequence, in separate messages.

The use of overlap is not supported on the NNI, as specified in ND1035 [1]

With regards to the UNI, NICC has published specifications for some particular applications:

- ND1033 [2] includes an option to use overlap
- ND1034 (SIP PBX UNI) [3] does not support overlap.

The use of overlap on a UNI is permissible as a Network Provider option but where it is used, conversion to enbloc must be completed prior to sending over an NNI.

Where conversion from overlap to enbloc involves variable length numbers appropriate measures should be taken to minimise delay.

5.2 How should Call Hold be handled?

Network Architecture Reference Points: A

Hold, and retrieval from hold, is signalled in SIP [5] using the directionality attribute of SDP as defined in RFC 3264 [4](section 8.4). NICC has noted that there are differing interpretations of how to construct an SDP offer to retrieve a call from hold and that in certain situations this can lead to it being impossible to resume two-way speech after a call is retrieved from being on hold.

NICC recommends that CPs implement access SBC logic to constrain SIP hold within the domain of a single customer. This logic involves the following steps for processing an SDP offer received from a customer UA to place a call on hold:

1. Store a local copy of the direction attribute received in the SDP offer (i.e. ‘sendonly’)
2. Set the direction attribute to “a=sendrecv” in the SDP offer that is sent forward to the network side
3. When processing the corresponding SDP answer, use the locally stored copy of the received direction attribute in the SDP offer to derive the direction attribute returned to the customer side in accordance with RFC 3264 [4] (i.e. ‘recvonly’).

Subsequently, when retrieving a call from hold, the customer UA will insert ‘sendrecv’ as the direction attribute and standard SDP offer/answer procedures will be followed.

Implementing at the access SBC guarantees interoperability both within and between CPs. Where a CP has not implemented this procedure, they must take steps to ensure that SIP hold is not signalled across NNIs.

5.3 Use of SIP OPTIONS 'ping'

5.3.1 Background

SIP is a connection-less protocol and has no standard for 'heartbeat' signalling between SIP entities. Therefore, the status of a SIP peer cannot be readily observed as used to be the case when using traditional telco (TDM-based) trunk links. However, devices in SIP networks (such as SBCs and proxies) still need an efficient means of forwarding calls to a next-hop device and without any such 'heartbeat' signalling, there is no way to tell in advance if a neighbouring device is able or willing to accept calls (which could mean calls are delayed or failed if the availability of a peer SIP entity is not determined until a call attempt is made).

A de facto standard for a 'heartbeat' signalling mechanism for SIP has emerged in the industry to address this problem. This standard utilises the SIP OPTIONS method between two devices; this mechanism is typically referred to as (SIP) OPTIONS "ping".

RFC3261 [5] defines the SIP OPTIONS method as a means to allow a SIP UA to query another SIP UA about its capabilities (such as supported methods, content-types, extensions and codecs). A SIP OPTIONS request may be used either inside or outside of an existing dialog. Although a SIP UA is not required to send SIP OPTIONS requests, a SIP UA is required to support the ability to respond to a SIP OPTIONS request if one is received.

Whilst the use of SIP OPTIONS "ping" provides a limited solution to determine the health of a peer SIP entity, the lack of any formal specification for using such a mechanism has led to inconsistent implementations amongst equipment manufacturers. This is despite the fact that RFC3261 [5] (in section 11.2) does state the type of response a SIP UA should generate when a SIP OPTIONS request message is received (i.e. "The response code chosen MUST be the same that would have been chosen had the request been an INVITE. That is, a 200 (OK) would be returned if the UAS is ready to accept a call").

5.3.2 SIP OPTIONS 'ping' across NNIs

Network Architecture Reference Points: C

NICC provides the following as guidance on the use of SIP OPTIONS as a 'heartbeat' / 'health-check' mechanism across Network-Network interfaces (NNIs):

For NNIs, if a SIP OPTIONS addressed to the next hop server is sent then the interpretation of any responses should be as follows:

- If no SIP OPTIONS response is received, the peer SIP entity should be deemed 'unreachable'; by implication, the peer SIP entity is therefore deemed to not be able to accept calls

- If a SIP OPTIONS response is received (regardless of the value of the response code value contained in it i.e. whether it is a 200 OK or otherwise), then the peer SIP entity is 'reachable' and is therefore may be able/willing to accept calls.

  Note: Any required interpretation of the specific response code that is contained in a SIP OPTIONS "ping" response message (e.g. the interpretation of received response code values in order to then further determine the ability and/or willingness of the peer to accept calls) will have to be subject to bi-lateral agreement between the involved CPs. NICC does not mandate or provide any guidance on what interpretation should be made of the response code values in SIP OPTIONS response messages

5.3.3 SIP OPTIONS 'ping' across UNIs

Network Architecture Reference Points: B

NICC does not provide any guidance on the use of SIP OPTIONS as a 'heartbeat' / 'health-check' mechanism across User-Network Interfaces (UNIs)

For UNIs, the use of SIP OPTIONS, and the interpretation of any responses must be by bi-lateral agreement between the parties involved.

5.4 Ring Tone Duration

Network Architecture Reference Points: A

One of the basic requirements for a voice call is that a call remains in the ringing phase for a minimum time of 90 seconds.

Some originating networks are known to have reduced this time in order to release network resources faster. However, this has been seen to have an adverse impact on some services. The prime example is Call Forward on No Reply (CFNR) where 30 seconds can elapse before the call is diverted. In the worst case observed, a call to a diverted line results in the called party's phone ringing for 4-5 seconds before being released.

ITU-T Q.118 [6] recommends a period of between 90 and 180 seconds be supported to ensure services such as CFNR are not impacted. SIP implementations in the UK should follow this.

5.5 Dealing with SIP Overload Control responses

Network Architecture Reference Points: B, C

ND1657 [7] is the definitive ND containing full details regarding overload control requirements for UK CPs across all external SIP interfaces (NNI and UNI).
The ND also states that the use of ND1657 in relation to other SIP interfaces is not precluded. It also notes that media congestion is not managed through SIP overload control.

The information contained in ND1657 applies to all UK networks, including SIP PBX implementations.

The key points from ND1657 include -

- The receipt of a 500 response code,
    - Indicates a significant portion of the network is congested
    - Indicates that the recipient network must not re-attempt the call, because to do so will likely amplify the congestion
    - Mandates the returning of 500 on each hop back to the originating network so that it plays Congestion tone to the caller.
- Priority is given to emergency calls
- There shall be no more than 6 attempts in total (per call) to route the call across any organisation’s egress nodes
- End user devices / phones shall not automatically re-attempt failed calls.

Annex <A> (normative): Items for further study

The following are items that may be considered for inclusion in this document in the future which require further study before they can be included.

- When are Preconditions required?
- Can I know that my call will be SIP end-to-end?
- When are Echo Cancellers used?
- Use of EISEC digits
- Backward media before answer
- Malicious Call ID
- SIP Response code for invalid number, e.g. 404, 484.

History

Document history		
Version	Date	Milestone
1.1.1	30th January 2024	Initial publication