HELSETJENESTENS DRIFTSORGANISASJON
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# SIP Trunk Requirements

SIP trunk using HDO siptrunk infrastructure

with multi-user configurations from Telenor IPT

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# 1. Scope

This document contains information on requirements put on customer premises equipment (CPE) for the SIP trunk to be compatible with HDO infrastructure and Telenor's IPT service.

The aim with the document is to guide system vendors, solution designers and system integrators when configuring the CPE, typically a pbx system or a control room environment.
The control room solution will be addressed as CPE in the document.

This specification will be revised as new methods and services are implemented.

The HDO SIP infrastructure currently consist of the following main building blocks:

- **The common HDO core** provide interconnection between health regions, HDO and different telecom operators, including Telenor IPT.
- **Session Border Controllers (SBC)** interfacing the CPE and providing a demarcation point.

Telenor IPT currently consist of the following main building blocks:

- **The application server ECN**, providing advanced fixed-mobile convergence services combining fixed and mobile services with switchboard, queues, voice recording etc.
- **The common IMS core** for all SIP UA's (business and consumer VOIP terminals plus mobile volte terminals)
- **Session Border Controllers (SBC)** interfacing HDO and providing a demarcation point

This specification describes the following scenario:

a) A multi-tenant SIP trunk terminated in HDO data center, where several control rooms use the same trunk.

B) Segregation and dedicated trunks for emergency calls (113) and other traffic.

# 2. Preface

## 2.1 General definitions

The key words 'MUST', 'MUST NOT', 'REQUIRED', 'SHALL', 'SHALL NOT', 'SHOULD', 'SHOULD NOT', 'RECOMMENDED', 'MAY', and 'OPTIONAL' are to be interpreted as described in RFC 2119 [1] an indicate requirement levels for compliant SIP implementations.

Calling Party /Caller: The client who *originates* the call

Called Party /Callee: The client who *receives* the call

## 2.2 Change management

HDO and Telenor will reserve the right to:

- Change this specification
- Change the SIP service in itself

## 2.3 SIP trunk model

HDO uses "Static Routing/Static Mode" which implies that the IP-PBX does not register towards HDO infrastructure. Instead, the IP-PBX has a peering connection with the Core network, and the routing paths are defined through provisioning:

Dynamic mode (SIP registration) is not supported.

## 2.4 Telenor IPT Partner

HDO are using a partner service from Telenor called IPT Partner that supports multi-user configuration. The term "PARTNER" is used for the hosting partner (HDO) who installs multi-tenant equipment on his premises, and aggregates traffic from the end-users (CPE) to a single SIP trunk/channel bank.

The PARTNER must make sure that all end-customers behave homogenous in respect of SIP methods/signalling.

### 2.4.1 IPT Partner Customer

PARTNER CUSTOMER is the end-customer, which subscribes to services from the PARTNER. In HDO's cases the partner customers are the different control rooms using the CPE solution.

### 2.4.2 Support for different CPE versions

CPE testing and approval is performed for a specific software/hardware version. As CPE software (SW) is continuously updated with improvements and bug fixes, Telenor will not require to test each new release for compatibility with IPT.

Telenor will state a minimum HW/SW version for different CPE's, any version newer than this is normally allowed.

SIP is not a fixed recommendation, changes are still being added and different vendors implement these additions in different pace. Interpretations of the SIP specs may also lead to functional challenges. Telenor can therefore not guarantee that all functions will work seamlessly between different SIP UA's.

Please note that all solutions have to be tested and ratified before put into service. As many PARTNER CUSTOMERs depend on the service, it is especially crucial that the perform well.

The HDO infrastructure are tested and approved by Telenor, however all solutions connecting to the HDO infrastructure need to be tested in accordance with the lates requirements before put into service. The testing is done together with HDO.

# 3. SIP Trunk Specifications

**NOTE!** Unless otherwise specified, all references to SIP are according to RFC3261.

## 3.1 Prerequisites for signalling and media handling

### 3.1.1 Multiple SBCs

HDO has multiple core systems working in parallel in order to provide a non-stop voice service. Each of these systems have duplicated and dedicated SBCs interfacing Customer solutions. These systems are located on different physical sites.

The SBC's have unique IP addresses, and CPE *must allow traffic* from all these addresses. The addresses are provided from HDO during installation.

- Outbound traffic (from Customers (CPE) to the SBC): The solution is free to choose how to balance traffic. It is recommended to use all available SBC IP addresses in a load balance configuration.
- Inbound traffic (from HDO to the customer (CPE)): The SBC will perform load balancing over the available Customer interfaces. Note that the traffic is balanced as a whole for all end user customers, hence a single customer may experience majority of traffic from one SBC.

### 3.1.2 Support for multiple CPE's

Up to 10 CPEs/IP addresses can be configured at the customer premises.
Distribution algorithms can be chosen between

- Round-robin
- Hunt
- Least Cost Routing (with Roud-Robin between equal costs)

Note that all CPEs must be capable to manage all E.164 numbers belonging to the customer / SIP trunk.

### 3.1.3 Addressing and routing

It is required that the signalling SIP layer IP-address and port received by HDO from the CPE is equivalent to the layer 3 IP-address and port used in the IP-header. (I.e. in case of a NAT between CPE and HDO, it must also translate the SIP payload.)

The HDO platform requires that the layer 3 (IP) address used for the signalling is fixed. This static address is provisioned in the HDO SBC.

### 3.1.4 Transport protocol

Preferred transport protocol is UDP. TCP is supported if we cannot use UDP.

#### 3.1.4.1 MTU size and fragmentation

Default MTU size is 1500 bytes. In some callcases the packet size might expand beyond this, so fragmentation should be supported.

Alternatively, TCP should be configured instead of UDP for SIP.

### 3.1.5 Port numbers

The platform uses port 5060 for SIP signalling for both UDP and TCP as default. Other signalling ports can be assigned if requested. The platform by default listens for RTP/RTCP on port 20000-25000 as signalled in SDP.

### 3.1.6 Portnumber and RTP Events

Network sockets (ip-adress/portnumber) SHOULD NOT be changed when sending RTP Events (e.g DTMF), as CPE on the B-side might not support this.

## 3.2 Multi-customer configuration

In order to separate the different control rooms on the SIP trunk, P-Preferred-Identity (PPI) header , defined in RFC 3325, MUST be used in INVITE messages. The value of the PPI must be unique for each control room. Failing to provide an identification will cause calls to fail.

The value assigned in the PPI will be discussed with the CPE provider during installation/Configuration of the CPE solution.

## 3.3 Numbering format and deviations

### 3.3.1 Standard number format

By default SIP-URI's are used for identifying E.164 telephone numbers.

Number format is international E.164 with "+" as the international prefix

```
sip:+CC<national_number>@ipt.hdo.com;user=phone
```

**Outgoing call from the CPE:**

The INVITE SHALL contain calling and called party number information as follows:

- Calling party must be included in From-header. Country Code MUST be included.
- Called party must be included in both Request URI and To-header. Country Code must be included.

**Incoming calls to the CPE:**

Country Code (+47) is always included in Called party (Request URI and To-header)

In case of a forwarded call (by the CPE) the SIP CPE MAY use a FROM number not assigned to the CPE. See paragraph 3.11.2

### 3.3.2 Emergency numbers

Emergency numbers (e.g. 110, 112 and 113) MUST be on the format +47xxx. The CPE SHALL not do any substitutions or permutations in the FROM number (caller/calling party number).

### 3.3.3 Deviations from standard number format

The service has defined some deviations from standard number format when PNP (Private Numbering Plan) is used:

In case of using the PNP service, the called PNP number should be signalled without international prefix or countrycode, example:

```
sip:2345@ipt.hdo.com;user=phone
```

where the PNP number = 2345.

## 3.4 INVITE from CPE

The figure attached in Appendix 1 shows an example of an invite from CPE to IPT where the various parts are discussed throughout this section.

It is recommended that the initial INVITE should include SDP offer, but it is not mandatory.

## 3.5 Use of Call Admission Control (CAC)

The number of simultaneous calls ("channels") on the SIP trunk is limited to the ordered and defined CAC value in order to prevent oversubscription of the IP connection carrying the SIP trunk. HDO and Telenor will reject an INVITE with 503 response code if number of channels are over seized.

## 3.6 Un-assigned numbers in CPE

INVITES from PSTN to customer numbers not in use, MUST be answered with appropriate response code (e.g 404 Not found). CPE MUST NOT route these calls back to PSTN, as this will cause routing loops and may block legitimate calls to/from the customer.

## 3.7 Audio codecs supported

Different codecs will be offered from the network, depending on the capabilities of the "other" side. G.711a is always offered, and MUST as a minimum be supported by CPE

SDP offers from the network containing codecs not supported by CPE, MUST be tolerated.

To ensure that HD voice is obtained when possible, the following codec order is recommended:

1. G.722. The network offers transcoding to/from HD voice -capable mobile terminals
2. G.711a

When responding to an initial codec offering, the CPE SHOULD reply with only 1 voicecodec as recommended in 3GPP IMS specification.

### 3.7.1 Silence suppression/comfort noise

Use of comfort noise (CN) for G.711 is supported. Payload type 13 is used in G.711/8 kHz clock rate. Telenor IPT PSTN media GW does not use silence suppression.

### 3.7.2 PackettTime (ptime) and maximum ptime

Recommended ptime value is 20 ms.

Maxptime = 40 ms

## 3.8 Early media

The CPE SHOULD support Early Media as defined in RFC3959/3960.

### 3.8.1 Calls to CPE

Telenor IPT in general blocks early media in backwards direction from CPE and back to caller.

For the calling party to generate/hear ring-back tone, a provisional response 180 Ringing MUST be sent from called party CPE

### 3.8.2 Calls from CPE

If Early Media is not supported, it may in some call scenarios result in loss of pre-connect tones and announcements (including ring-back tones).

### 3.8.3 Support for PRACK (Reliability of Provisional Responses)

HDO dose not make use of PRACK for outgoing calls towards Telenor IPT the reason for this, is that local ring-back tone will not be generated in some call cases when PRACK is in use.

If CPE signals support for PRACK but not UPDATE, HDO will not make use of PRACK (by not sending "Required: 100rel" in provisional respons.
Telenor requires support for UPDATE together with PRACK due to some call-forward and no-reply scenarios.

## 3.9 Keep alive session timers

The CPE SHOULD support RFC 4028 (Session Timers in the Session Initiation Protocol)

Value for Session Expires timer is recommended to be 1800 seconds if session refreshment is supported.

If session refreshment is **not** supported, the Support Header MUST NOT contain the parameter Timer.

## 3.10 Presentation restrictions (privacy)

### 3.10.1 Calls terminating on the CPE

When a calling party number has presentation restrictions, the HDO service will insert userpart ”anonymous” in the From-header towards CPE.

When calling party number is not available to Telenor IPT or HDO, the HDO service will insert userpart=”unknown” in the From-header towards CPE.

### 3.10.2 Calls initiated by the CPE

Preferred method for user provided privacy is for CPE to send "anonymous@anonymous.invalid" in the From-header.

Alternatively, the CPE may use Privacy header: "user". This can be combined with either sending the originating E.164 number or with using "anonymous@anonymous.invalid" in the From-header.

In both cases the IPT privacy service will be similar to the use of "CLIR" in ISDN environments.

### 3.10.3 Privacy override

Emergency Calls coming from Telenor IPT to HDO infrastructure are delivered on a separated sip trunk that have OIR override. The CPE solution must support having dedicated trunks for emergency calls, to receive calls that do not have presentation restrictions.

The header P-Asserted-Identiy will be included and consist of the callee's phone number. The number are also signaled in the from header, and the privacy header are included with the value 'user;id'

If a call are received by the CPE that are tagged with privacy override, the number must be hidden if the call are transferred out towards the PSTN network or other control rooms that are not allowed to see the callee's number.

Appendix 2 in this document are showing a signalling example.

## 3.11 Call Forwarding

### 3.11.1 By using 302 Moved Temporarily

In case the CPE performs Call Forwarding, it can be done by means of tromboning[1] or by responding on the INVITE with a 302 Moved Temporarily. In this case the C-party number (full international E.164 number) must be put in the user part of Contact header.

Below an example of a 302 SIP message is shown.

```
Request Line
   SIP/2.0 302 Moved Temporarily
Headers
   Via: SIP/2.0/UDP 148.122.250.11:5060;branch=z9hG4bKjq1oak3088jhef4de640.1
   From: <sip:+4795123628@148.122.250.11:5060;user=phone>;tag=SDek3fa01-52049bc8-13995481
   To: <sip:+4722391267@ipt.telenor.com:1035;user=phone>;tag=2768870514
   Call-ID: SDek3fa01-c2aa6c2e2f
   CSeq: 1 INVITE
   Allow: INVITE, ACK, CANCEL, BYE, NOTIFY, REFER, OPTIONS, UPDATE, PRACK, SUBSCRIBE, INFO
   Allow-Events: talk, hold, conference, LocalModeStatus
   Contact: <sip:+4767537980@ipt.telenor.com>
   Server: IPT Test CPE 2.0
   Supported: path
   Content-Length: 0
```

If forwarding is performed by the CPE, then the History-info header should be included in the outgoing INVITE

### 3.11.2 Call forwarding: Use of History-Info

When a call is forwarded by an IP-PBX, the From-header usually contains the original called number so the receiver of the forwarded call will see who is actually calling. In this case, Telenor will by default charge the main number of the customer. History-Info should be used to indicate the *forwarding-by* number, which must be within the defined number range of the customer. This number will also be used for charging if valid.

The History-info header MUST include an index and should also include a SIP response cause

The format is as follows for a call forwarded from +4722391111 to +4722392222:

```
History-Info: <sip:+4722391111@ipt.telenor.com?Reason=SIP%3Bcause%3D302>;index=1
History-info: <sip:+4722392222@ipt.telenor.com;user=phone;cause=302>;index=1.1
```

[1] An incoming call to a user for whom forwarding is activated on the CPE, shall result in the CPE setting up an outgoing call and trombone the call signalling between the 2 calls. Media from the 2 calls may be joined internally on the CPE or the CPE may have logic for anti-tromboning handing off the media to the Telenor IPT platform

### 3.11.3 Call forwarding: Use of Diversion header

It is recommended to use History-Info and not Diversion since the latter is an old standard defined in a Historical RFC. However diversion are supported for now.

## 3.12 Call Hold/Resume

Call Hold / Resume is performed using methods according to RFC3264 (SDP Offer/Answer).

## 3.13 Call Transfer using REFER

Call Transfer is performed using methods according to RFC3264 (SDP Offer/Answer).

The REFER method can be used for call transfer on IPT. REFER is supported for both blind transfer and attended transfer. REFER is supported for the call scenario where a call is initiated from an external number towards the CPE, and the CPE transfer the callee to a 3rd party number. The format of the Refer-To: field is:

- ❑ For blind transfer, refer to a telephone number in SIP URI:
  `Refer-To: sip:+4722391267@ipt.telenor.com;user=phone`
- ❑ For consult transfer, refer to a phone number and a call ID, for example for call-ID 01234abc:
  `Refer-To: sip:+4722391267@ipt.telenor.com;user=phone?Replaces=01234abc`

Please note that REFER is an add-on product, and by default not enabled.

## 3.14 Call Waiting

There is no specific functionality to support Call waiting indication. Call waiting functionality will be according to functionality implemented in the CPE.

## 3.15 DTMF

2 options exist:

- ❑ Recommended solution for in- and outgoing DTMF is according to RFC2833 or the newer RFC 4733
- ❑ If RFC 2833/4733 is not supported and the CPE signals support for SIP INFO, Telenor IPT will send to and accept from CPE DTMF using SIP INFO with "content type = application/dtmf-relay".

SIP INFO with "content type = audio/telephone-event" is NOT supported.

DTMF inband might work, but this method is vulnerable to jitter and might be corrupted in a scenario with jitter buffers.

See also chapter 3.1.6, use of RTP Events and portnumbers.

## 3.16 Fax operation

Generally, T.38 fax mode is NOT supported.

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# 4. References and abbrevations

| Document links | Description |
|---|---|
| RFC 2119 | Defined key words used in IETF RFCs (and in this document) to indicate requirement levels |
| RFC 3261 | Main SIP RFC specification |
| RFC 3264 | Describes the negotiation mechanisms for media sessions using SDP. |
| RFC 2833 | Describes the RTP payload for DTMF digits, fax and other telephony tones |
| RFC 4028 | Describes SIP session timers |
| RFC 4733 | Describes the RTP payload for DTMF digits, fax and other telephony tones |
| RFC 3960 | Describes how to manage early media in SIP using two models: the gateway model and the application server model |
| RFC 3959 | Defines a new disposition type (early-session) for the Content-Disposition header field in SIP |
| RFC 3311 | Defines the new UPDATE method for SIP |
| RFC 3262 | Specifies an extension to SIP providing reliable provisional response messages |
| RFC 5806 | Historic RFC which describes the DIVERSION header, that contains information on diverted or forwarded (ISUP redirected) PSTN call, so that the original party information is still available. |
| T.38 standard | Standard describing facsimile communication over IP networks. |
| RFC 2976 | The SIP INFO Method |
| RFC 3581 | An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing |
| RFC 3515 | The SIP REFER Method |
| RFC 7044 | An Extension to the Session Initiation Protocol (SIP) for Request History Information |

| Abbreviation | Meaning |
|---|---|
| 3GPP | 3rd Generation Partnership Project, Standardizing body, originally for 3G mobile services but later extended to LTE, 4G and IMS |
| AMR-WB | High quality voice codec used with mobile terminals. Also referred to as G.722.2 |
| CAC | Call Admission Control. Limits the numbers of simultaneous calls ("channels") on the SIP trunk in order to prevent oversubscription of the IP connection. |
| CLIR | Calling Line Identification Restriction. |
| CPE | Customer Premise Equipment. A general term for different customer placed equipment as PBX. |
| DSL | Digital Subscriber Line. General term for transmission technology over copper based access circuits. |
| DTMF | Dual Tone Multi Frequency. Tone signalling with phone keyboard for voicemail, IVR etc. |
| E.164 | An ITU standard defining the international numbering format, e.g. +4712345678 representing a Norwegian 8 digit number, country code (47) and international prefix (+ or 00) |
| ECN | Enterprise Communication Network. The application server that produce the MBN service |
| ESS | A Telenor database for mapping E.164 numbers with physical addresses used by emergency agencies |
| G.711 | Standard voice codec used in digital transmission network |
| G.722 | High quality voice codec used with fixed terminals. |
| IMS | IP Multimedia Subsystem. A standard originally defined by 3GPP how to produce multimedia services over a packet based system. Covers both mobile and fixed terminals. |
| MBN | Mobilt Bedriftsnett. Telenor hosted centrex offering covering both mobile, ISDN/PSTN and SIP terminals |
| MPLS-VPN | Virtual Private Networks based on Multi-Protocol Layer Switching. A standard for providing VPN's for customers. Allows use of private numbering. |
| NAT | Network Adress Translation. Used for mapping private addresses behind a firewall to public external addresses. |
| PBX | Privat Branch eXchange. An internal phone system with lines to the public network |
| PNP | Privat Numbering Plan. Giving the end-user the possibility to use short numbers |
| PRACK | Provisional Respons Acknowledgement. A standard for creating reliable responses in IMS networks. |
| RFC | Request for Comment, an Internet Engineering Task Force memorandum on Internet standards and protocols |
| RTCP | RTCP stands for Real-time Transport Control Protocol and is defined in RFC 3550. RTCP works hand in hand with RTP. RTP does |

| Abbreviation | Meaning |
|---|---|
| | the delivery of the actual data, whereas RTCP is used to send control packets to participants in a call. |
| RTP | Real Time Protocol.  A commonly used transport protocol for real-timer communication. |
| SBC | Session Border Controller. A network element performing several tasks in a SIP environment:  Topology hiding, perimeter protecition, correction of SIP parameters etc. |
| SDP | Session Description Protocol is used for describing multimedia communication sessions for the purposes of session announcement, session invitation, and parameter negotiation. |
| SIP | Session Initiated Protocol.  An IETF standard describing peer-peer multimedia communication over IP networks |
| SIP UA | SIP User Agent.  This is the network element where the SIP protocols are implemented, and represents the entity where SIP is translated |
| SIP-URI | A SIP URI is the SIP addressing schema, or identifying string of characters, to call another person via SIP. It is, essentially, a user's sip "phone number," and it is in a format similar to email. The format is "sip:user@host" |
| TCP | Transport Control Protocol is a standard that defines how to establish and maintain a network conversation via which application programs can exchange data. It is used for establishing reliable connections. |
| UDP | User Datagram Protocol.  Transport protocol used for low-latency, fault tolerant connections as an alternative to TCP,  Offers no error correction or re-transmission. |
| VOIP | A general term for Voice Over IP |
| | |

# Appendix 1: Outgoing INVITE

```
INVITE                                 : INVITE sip:+4767537980@148.122.250.150:5060 SIP/2.0
  --- Invite Method Headers ---
  Via                                  : Via: SIP/2.0/UDP 172.19.1.117:5060;branch=z9hG4bKEC9A6C8
  Remote-Party-ID                      : Remote-Party-ID: <sip:+4766718833@172.19.1.117>;party=calling;screen=yes;privacy=off
  From                                 : From: <sip:+4766718833@172.19.1.117>;tag=4397D020-22A3
  To                                   : To: <sip:+4767537980@148.122.250.150>
  Date                                 : Date: Thu, 06 Dec 2018 13:20:44 GMT
  Call-ID                              : Call-ID: 923E562D-F89011E8-9330F9FD-EC254672@172.19.1.117
  Supported                            : Supported: 100rel,timer,resource-priority,replaces,sdp-anat
  Min_SE                               : Min-SE:  1800
  Unknown Header                       : Cisco-Guid: 2684601344-0000065536-0000003337-1929450412
  User-Agent                           : User-Agent: Cisco-SIPGateway/IOS-12.x
  Allow                                : Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
  CSeq                                 : CSeq: 101 INVITE
  Timestamp                            : Timestamp: 1544102444
  Contact                              : Contact: <sip:+4766718833@172.19.1.117:5060>
  Expires                              : Expires: 180
  Allow Events                         : Allow-Events: telephone-event
  Max-Forwards                         : Max-Forwards: 69
  Session Expires                      : Session-Expires:  1800
  Content-Type                         : Content-Type: application/sdp
  Content-Disposition                  : Content-Disposition: session;handling=required
  Content-Length                       : Content-Length: 303
  CRLF
    === Session Description Protocol ===
    Protocol Version                   : v=0
    Session Owner/Creator and session identi : o=CiscoSystemsSIP-GW-UserAgent 4648 9909 IN IP4 172.19.1.117
    Session Name                       : s=SIP Call
    Session Connection Data            : c=IN IP4 172.19.1.117
    Time the session is active         : t=0 0
    Media Name                         : m=audio 17446 RTP/AVP 8 18 98 19
      Media                              : audio
      Port                               : 17446
      Protocol type                      : RTP/AVP
      Media Format                       : 8 = PCMA
      Media Format                       : 18 = G729
      Media Format                       : 98 = dynamic
      Media Format                       : 19 = reserved
    Media Connection Data              : c=IN IP4 172.19.1.117
    Media Attribute                    : a=rtpmap:8 PCMA/8000
    Media Attribute                    : a=rtpmap:18 G729/8000
    Media Attribute                    : a=fmtp:18 annexb=no
    Media Attribute                    : a=rtpmap:98 telephone-event/8000
    Media Attribute                    : a=fmtp:98 0-16
    Media Attribute                    : a=rtpmap:19 CN/8000
```

# Appendix 2: Incoming INVITE with OIR override

INVITE : INVITE sip:+4722391591@10.233.200.28:5060;user=phone SIP/2.0
--- Invite Method Headers ---
Via : Via: SIP/2.0/UDP 148.122.98.10:5060;branch=z9hG4bKibu11m003o3n42gjtd80.1
To : To: <sip:+4722391591@ipt.telenor.com:5060;user=phone>
From : From: "+4795123628" <sip:+4795123628@ipt.telenor.com:5060>;tag=SD7qfsa01-p65539t1622013141m452107c586s1_1562260705-1259046866
Call-ID : Call-ID: SD7qfsa01-794a04d82a02f784fc3ca6348aca9259-v300g00040
CSeq : CSeq: 1 INVITE
Max-Forwards : Max-Forwards: 65
Content-Length : Content-Length: 671
Contact : Contact: <sip:p65539t1622013141m452107c586s1@148.122.98.10:5060;transport=udp>;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Content-Type : Content-Type: application/sdp
Allow : Allow: REGISTER, REFER, NOTIFY, SUBSCRIBE, PRACK, UPDATE, INVITE, ACK, OPTIONS, CANCEL, BYE
Accept : Accept: application/sdp
Accept-Contact : Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Supported : Supported: timer, 100rel, histinfo
P-Assert-Identity : P-Asserted-Identity: <sip:+4795123628;cpc=ordinary@ipt.telenor.com;user=phone>
Min_SE : Min-SE: 900
Session Expires : Session-Expires: 1800
Privacy : Privacy: user;id
Max-Breadth : Max-Breadth: 10
Session ID : Session-ID: 1536ee99c56c9625189bfc9bbc