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*Technical Specification*

**Universal Mobile Telecommunications System (UMTS);  
LTE;  
Session Initiation Protocol (SIP) based user configuration;  
Stage 3  
(3GPP TS 24.238 version 8.2.0 Release 8)**

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# Foreword

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  - 1 presented to TSG for information;
  - 2 presented to TSG for approval;
  - 3 or greater indicates TSG approved document under change control.
- y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.
- z the third digit is incremented when editorial only changes have been incorporated in the document.

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# 1 Scope

The present document provides a Session Initiation Protocol (SIP) based protocol framework that serves as a means of user configuration of supplementary services in the IP Multimedia (IM) Core Network (CN) subsystem. The protocol framework relies upon the contents of the Request-URI in a SIP INVITE request to enable basic configuration of services without requiring use of the Ut interface.

The present document is applicable to User Equipment (UE) and Application Servers (AS) which are intended to support user configuration of supplementary services.

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# 2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

- [1] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".
- [2] 3GPP TS 24.173: "IMS multimedia telephony communication service and supplementary services; Stage 3".
- [3] 3GPP TS 24.229: "IP Multimedia Call Control Protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3".
- [4] RFC 4967 (July 2007): "Dial String Parameter for the Session Initiation Protocol Uniform Resource Identifier".
- [5] RFC 3966 (December 2004): "The tel URI for Telephone Numbers".

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# 3 Definitions, symbols and abbreviations

## 3.1 Definitions

For the purposes of the present document, the terms and definitions given in 3GPP TR 21.905 [1] and the following apply. A term defined in the present document takes precedence over the definition of the same term, if any, in 3GPP TR 21.905 [1].

## 3.2 Abbreviations

For the purposes of the present document, the abbreviations given in 3GPP TR 21.905 [1] and the following apply. An abbreviation defined in the present document takes precedence over the definition of the same abbreviation, if any, in 3GPP TR 21.905 [1].

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# 4 SIP based user configuration

## 4.1 General description

SIP-based protocol framework serves as a means of user configuration of supplementary services in the IM CN subsystem specified in 3GPP TS 24.173 [2]. The contents of the Request-URI in a SIP INVITE request is used to convey the configuration code to the Application Server that hosts the supplementary service. The "user=" parameter within the Request-URI is set by the UE to value "dialstring" as described in RFC 4967 [4]. Upon session initiation, the contents of the Request-URI are delivered by means of normal session setup signalling, as described in 3GPP TS 24.229 [3] to an Application Server. The Application Server then acts upon the Request-URI contents to effect the desired configuration data change (e.g., register and activate Communication Forwarding unconditional).

## 4.2 Syntax requirements

The precise digit sequences within the Request-URI that comprise the effective dialstrings for user configuration are defined by the IM CN subsystem service provider and are not subject to standardisation.

NOTE 1: The digit sequence corresponding to the feature code can begin with a special character such as "#" or "\*" according to network operator preferences. The length of the digit sequence is also defined by the network operator.

The digit sequences corresponding to the feature code shall be transported to the AS in the Request-URI of a SIP INVITE request as follows:

- as a SIP URI dial string conforming to RFC 4967 [4] where the "phone-context" parameter is set to the home network domain name and the "user" parameter is set to "dialstring";
- as a SIP URI that is not a GRUU and with the user part starting with a + and the "user" parameter equals "phone". The domain is set to the home network domain; or
- as a tel-URI with a "phone-context" parameter set to the home network domain as defined in RFC 3966 [5].

## 4.3 Signalling requirements

### 4.3.1 General

Two roles are recognized for the implementation of SIP-based user configuration:

- 1) UE (SIP-based user configuration client); and
- 2) Application Server.

### 4.3.2 Actions at the originating UE (SIP-based user configuration client)

When performing SIP-based user configuration, the UE shall create a SIP URI, as described in RFC 4967 [4], with:

- a) a dialstring, set to either the concatenation of feature code and the number to be provisioned or the feature code alone if no number information needs to be provided for the service; and
- b) a "phone-context" parameter, set to the home network domain name.

The UE shall construct and initiate an appropriate INVITE in accordance with 3GPP TS 24.229 [3] with the Request-URI set to the URI created above.

### 4.3.3 Actions at the AS serving the originating UE

Upon receiving an INVITE request with Request-URI containing a SIP URI including a 'phone-context' parameter equal to the home network domain name, a "user=" parameter set to value "dialstring", and a dialstring representing a feature code or a concatenation of feature code and number to be provisioned, the AS shall:

- a) accept the INVITE request following the procedures specified for AS acting as a terminating UA in 3GPP TS 24.229 [3]; and
- b) perform service activation, deactivation, or configuration data modification based on the recognized contents of the Request-URI.

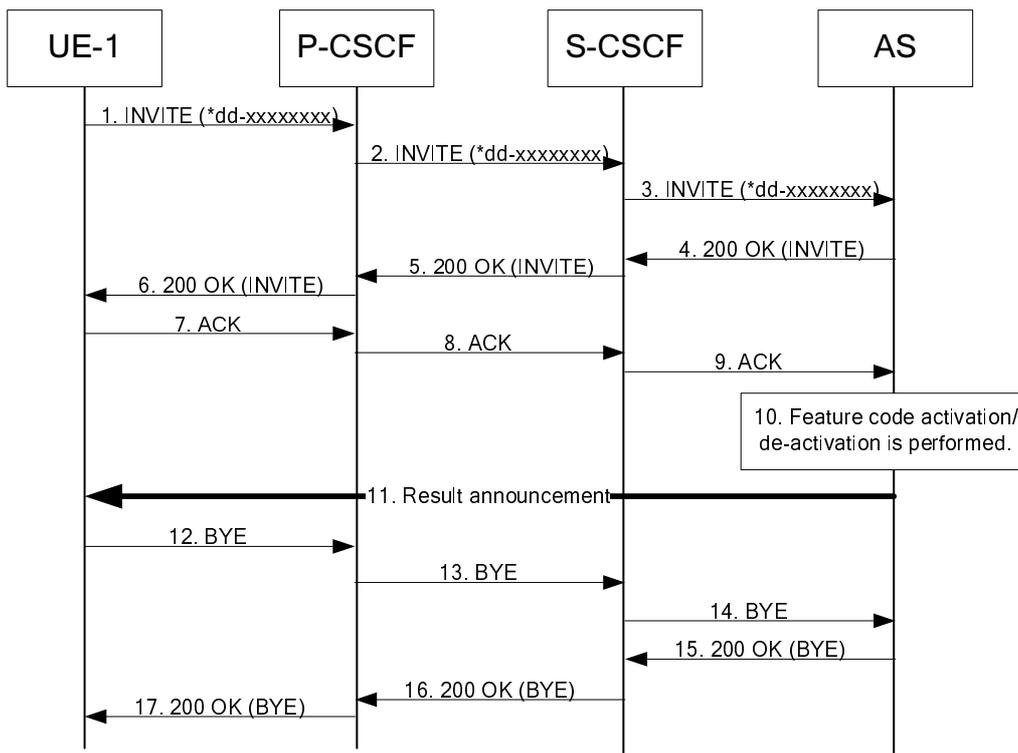
Based on the outcome of the service configuration operation, the AS may play an appropriate announcement to notify the user of the result of the operation.

NOTE: The Request-URI of the received INVITE request can contain a SIP URI with the "user=" parameter set to value "phone" or a Tel-URI. The AS can treat this Request-URI as a dialstring.

# Annex A (informative): Signalling flows

## A.1 User provisioning by SIP-based user configuration

The signalling flow below illustrates the use of the SIP-based user configuration capability. This basic capability is used for activation, deactivation and configuration data modification.



**Figure A-1: User provisioning using feature code**

1-9. UE-1 establishes a call with the provisioning AS following normal call setup procedure. UE-1 sends the provisioning information as part of the Request URI.

**Table A.1-1: INVITE request (UE-1 to P-CSCF)**

```

INVITE sip:*12345;phone-context=home1.net;user=dialstring SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>
Accept-Contact: *;+g.3gpp.icsi_ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel"
P-Preferred-Identity: "John Doe" <sip:user1_public1@home1.net>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>;tag=171828
To: < sip:*12345;phone-context=home1.net;user=dialstring>
Call-ID: cb03a0s09a2sdfg1kj490333
Cseq: 127 INVITE
Require: sec-agree
Supported: precondition, 100rel, gruu
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg= hmac-sha-1-96; ealg= aes-cbc; spi-c=98765432; spi-s=87654321; port-c=8642; port-s=7531
Contact:
<sip:user1_public1@home1.net;gr=hdg7777ad7aflzig8sf7>;comp=sigcomp;+g.3gpp.icsi_ref="urn%3Aurn-7%3gpp-service.ims.icsi.mmtel"
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
    
```

```
Accept: application/sdp; application/3gpp-ims+xml
Content-Type: application/sdp
Content-Length: (...)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=inactive
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

**Request-URI:** in this example, the configuration feature code is set to \*12345 and is represented as a dialstring.

**SDP:** the SDP is included for audio media, facilitating possible usage of audio announcements, DTMF tones and IVR interaction depending on how the network operator deploys the service configuration.

10. The AS performs feature activation, deactivation or configuration data modification based on the information received from the UE.

11. The AS can also, by interaction with an MRFC, announce the result of the configuration operation to UE-1.

12-17. After the provisioning is completed, UE-1 sends a BYE request to terminate the call with the AS.

## Annex B (informative): Change history

Change history							
Date	TSG #	TSG Doc.	CR	Rev	Subject/Comment	Old	New
2008-04					Version 0.0.0 TS Skeleton	0.0.0	
2008-04					C1-081430: " Scope and General description for SIP based user configuration"		0.1.0
2008-04					C1-08146: " Text for syntax and signalling requirements for SIP based user configuration"		0.1.0
2008-04					C1-081112: " Example signalling flows for SIP based used configuration"		0.1.0
2008-05					Implemented C1-081948, C1-082046		0.2.0
2008-07					Implemented C1-082447 from CT1#54		0.3.0
2008-08					Implemented C1-082958, C1-083203, C1-083396		0.4.0
2008-09					Version 1.0.0 created for presentation to TSG CT#41 for information	0.4.0	1.0.0
2008-11					Miscellaneous editorial cleanups.	1.0.0	1.0.1
2008-11					Version 2.0.0 created for presentation to TSG CT#42 for approval	1.0.1	2.0.0
2008-12	CT#42				Version 8.0.0 created after approval in CT#42	2.0.0	8.0.0
2009-03	CT#43	CP-090121	000	1	Correction of URN-value for Service Identifiers	8.0.0	8.1.0
2009-12	CT#46	CP-090908	000	4	Removal of editor's note from 24.238	8.1.0	8.2.0

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## History

<b>Document history</b>		
V8.0.0	January 2009	Publication
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