- SIPT - Test Suite development sample -

This document describes the successive development steps to build a SIP-T Executable Test Suite.

The development is based on some standard products and it uses large parts of code from existing ETS:

- ETS 300 356-33: ISUP-V3 Basic Call
- VoIP SIP ETS – rfc3261
- VoIP ISUP SIP ETS – rfc 3398

The objective is to demonstrate:

- creation of the ATS TTCN mp source file from existing protocol definitions
- compilation of the ATS and development of customisation files required to interface the ETS to the Tester
- test of a few simple Test Cases using a basic IUT

1. ATS Definition

1.1 ATS references

Interface under test specifications	SIP-ISUP gateway
Protocol specifications	SIP RFC 3372 SIP-T SIP RFC3261 and ISDN User Part ITU-T Recommendation Q.784.2 (1997) RFC 3398
ATS specifications	Based on test purpose samples as defined hereafter
ATS file name	SIP-T

1.2 Declarations Part

1.2.1 RFC 3398 Declarations parts

The Declaration Part of the ETS that was previously developed for testing Sip-ISUP mapping from rfc3398 has been used as a starting template. The Declaration Part include Declarations about the ISUP and the SIP protocol, but it does not support the encapsulation of ISUP in SIP message as required by rfc3372.

1.2.2 Components of the SIP-T Protocol

1.2.2.1 Core SIP

Rfc3372 5.1:

```
SIP-T uses the methods and procedures of SIP as defined by RFC 3261.
```

Types, ASP, PDU are defined to support rfc3261. New SIP messages are defined to include an ISUP body, as initially only an SDP body was supported.

PDU Type Definition	
PDU Name:	SIP_REQUEST_IAM
Group:	
PCO Type:	SIP_TRANSPORT
Encoding Rule Name:	
Encoding Variation:	
Comments:	SIP REQUEST Message RFC3261 SIP-message = Request | Response

Field Name	Field Type	Type Encoding	Comments
request_line	Request_Line_type		
headers_str	IA5String		Large String including all headers fields
headers	Message_header_type		
body	IA5String		
ISUP_body	ISUP_IAM		ISUP IAM PDU

Detailed Comment:	

The **body** field is used for SDP body, and an **ISUP_body** field has been added to carry the ISUP message.

The MIME decoding is achieved by the Osip library. The MIME headers are programmed by PIXITs and they have not been defined in type definition: this could be a future improvement.

1.2.2.2 Encapsulation:

Rfc3372 5.2:

```
Encapsulation of the PSTN signaling is one of the major requirements
of SIP-T.  SIP-T uses multipart MIME bodies to enable SIP messages to
contain multiple payloads (Session Description Protocol or SDP [5],
ISUP, etc.).  Numerous ISUP variants are in existence today; the ISUP
MIME type enable recipients too recognize the ISUP type (and thus
determine whether or not they support the variant) in the most
expeditious possible manner.  One scheme for performing ISUP
encapsulation using multi-part MIME has been described in [2].
```

The ISUP messages can be encapsulated as simple body or using multipart MIME bodies.

The received SIP message will be decoded in both cases by the SIP parser, and the same Constraint definition can be applied: in the case of a simple ISUP body, SDP body is considered to be omitted, and in the case of a simple SDP body ISUP body is considered to be omitted. Processing MIME decoding by the parser library avoids multiple definitions of the SIP message to support the miscellaneous encoding variations: simple or multipart body, order of the bodies, …

The sent SIP message is processed by the Send Functions of the customisation files, that are in charge of SIP message encoding. Depending on content-type header, multi-part MIME encoding will be used.

Some PIXITs have been added to define ISUP body header lines:

Test Suite Parameter Declarations

Group:

Parameter Name	Type	Default Value	PICS/PIXIT Ref	Comments
TSP_MULTIPART_BOUNDARY	IA5String			Multipart body - Unique boundary
TSP_MULTIPART_SDP_CONTENT_TYPE	IA5String			Multipart body - SDP Content-Type header line
TSP_MULTIPART_ISUP_CONTENT_TYPE	IA5String			Multipart body - ISUP Content-Type header line
TSP_MULTIPART_ISUP_CONTENT_DISPOSITION	IA5String			Multipart body - ISUP Content-Disposition header line

Detailed Comment:

Example of ISUP message / simple ISUP body:

```
      16:54:13/146.9 3-     D4   +D4  DATA
                                  UNIT--------------------------------[00661]+
                                 ¦  Protocol: UDP        from IP             ¦
                                 ¦  S:172.20.73.230    D:172.20.1.177        ¦
                                 +-------------------------------------------+
                    D4  UDP   (S:05060 D:05060)  Len:00627 Chk:72ef [0619]
                    D5  SIP                                         [0619]
                        SIP/2.0 180 Ringing
                        Via: SIP/2.0/UDP172.20.1.177:5060;branch=z9hG4bK25727
                        To: <sip:29612609876@italtel.com:5060;user=phone>;tag=
                        From: "ClarinetUserDisplay" <sip:111111111@172.20.1.17
                        Call-ID: 779528506@172.20.1.177
                        CSeq: 10 INVITE
                        Supported:  100rel
                        P-Early-Media:  direction=sendrecv
                        Accept: application/sdp, application/isup, application
                        Max-Forwards:  70
                        Contact: <sip:29612609876.iIiIiI.ac14cc27.@172.20.73.2
                        Content-Type: application/isup; version=SIP; base=itu-
                        Content-Disposition: signal; handling=optional
                        Content-Length: 8

                        06 16 14 01 29 01 01 00
                        D6  SIP-T ISUP/ACM (Address complete)
                        (--) len:-- >> Backward call indicators
                        Oct 1 : 00------ End-end method    = Not available
                        --01---- Called category  = Ordinary subscrib
                        ----01-- Called status    = Subscriber free
                        ------10 Charge indicator = Charge
                        Oct 2 : 00------ SCCP method  = No indication
                        --0----- Echo control = Incom half not includ.
                        ---1---- ISDN access  = Termin. access ISDN
                        ----0--- Holding      = Not required
                        -----1-- ISUP indic.  = Used all the way
                        ------0- End-end info.= No info. available
                        -------0 Interworking = Not encountered
                        (--) len:-- >> Pointer to start of optional part : 01
                        (29) len:01 >> Optional backward call indicators
                        Oct 1 : ----0--- MLPP indic.  = No indication
                        -----0-- Segmentation = No addition. info
                        ------0- Call divers. = No indication
                        -------1 In-band info = Available
                        (00) len:00 >> End of optional parameters
```

Example of SIP message / multi-part body:

```
    11:00:51/078.4 3-     D4  +D4  DATA UNIT---------------------- [01066]+
                               |  Protocol: UDP        from IP            |
                               |  S:172.20.73.230    D:172.20.1.177       |
                               +------------------------------------------+
                    D4  UDP   (S:05060 D:05060)  Len:01032 Chk:e6ef     [1024]
                    D5  SIP                                             [1024]
                        INVITE sip:1774567@172.20.1.177;user=phone SIP/2.0
                        Via: SIP/2.0/UDP 172.20.73.230:5060;branch=z9hG4bK.iI
                        To:  <sip:1774567@172.20.1.177;user=phone>
                        From: <sip:unavailable@hostportion>;tag=01d768bc
                        Call-ID: 01d768e901d768d-2@172.20.204.38
                        CSeq:  1 INVITE
                        Route: <sip:172.20.1.177;lr>
                        Max-Forwards: 70
                        Contact: <sip:anonymous.iIiIiI.ac14cc26.@172.20.73.23
                        Allow: INVITE, ACK, PRACK, CANCEL, BYE, OPTIONS, MESS
                        Supported: 100rel
                        Accept: application/sdp, application/isup, applicatio
                        Content-Type: multipart/mixed; boundary=unique-bounda
                        MIME-Version: 1.0
                        Content-Length: 350

                        --unique-boundary-1
                        Content-Type: application/sdp

                        v=0
                        o=- 2551116 0 IN IP4 10.194.118.140
                        s=IMSS
                        c=IN IP4 10.194.118.140
                        t=0 0
                        m=audio 4010 RTP/AVP 8
                        b=AS:64

                        --unique-boundary-1
                        Content-Type: application/isup; version=FTSSURI; base
                        Content-Disposition: signal; handling=required

                        01 00 40 01 0a 02 02 08 06 83 10 71 47 65 07 08
                        01 00 00
                        --unique-boundary-1--
                        D6  SIP-T ISUP/IAM (Initial address message)
                        (--) len:-- >> Nature of connection indicators
                        Oct 1 : ---0---- Echo ctrl   = Half echo not included
                        ----00-- Cont. check = Not required
                        ------00 Satellite   = No circuit
                        (--) len:-- >> Forward call indicators
                        Oct 1 : 01------ ISUP pref.  = Not req. all the way
                        --0----- ISUP indic. = Not used all the way
                        ---0---- End-end inf = Not available
                        ----0--- Interwork.  = Not encountered
                        -----00- Method. ind = No method available
                        -------0 Call indic. = as National call
                        Oct 2 : -----00- SCCP method = No indication
```

As the ISUP routing_label and circuit_identification_code are not part of the ISUP message body in the SIP message, the ISUP PDU types have been modified to allow using the same body definition in both SIP and SS7 PDUs.

For example, the Tabular PDU Type definition for ISUP ACM was defined as follow:

PDU Type Definition

PDU Name:	ACM
Group:	
PCO Type:	ISUP_PCO
Encoding Rule Name:	
Encoding Variation:	
Comments:	Address complete (TABLE 21 / Q.763)

Field Name	Field Type	Type Encoding	Comments
<-	routing_label		m
<-	circuit_identification_code		m
MType	message_type		m
BCI	backward_call_indicators		m
opt_part_ptr	pointer		m
OBCI	optional_backward_call_indicators		o
CRef	call_reference		o @
Cause	cause_indicators		o
UUInd	user_to_user_indicators		o
UUInf	user_to_user_information		o
ATP	access_transport		o
GenNot	generic_notification_indicator		o 1.
TMU	transmission_medium_used		o
EchoInf	echo_control_information		o
ADInf	access_delivery_information		o
RnNb	redirection_number		o
ParCmp	parameter_compatibility_information		o
CDInf	call_diversion_information		o
NtwFac	network_specific_facility		o @
RemOp	remote_operations		o @
ServAct	service_activation		o @
RnNbRes	redirection_number_restriction		o
ConTrInd	conference_treatment_indicators		o
UIDAcInd	UID_action_indicators		o
EndOP	end_of_optional_parameters_indicator		o

Detailed Comment:	1. This parameter could be included several times. @ For national use only Note: The order of the optional parameters (o) can be arbitrary.

The definition has been changed to:

PDU Type Definition	
PDU Name:	ACM
Group:	
PCO Type:	ISUP_PCO
Encoding Rule Name:	
Encoding Variation:	
Comments:	

Field Name	Field Type	Type Encoding	Comments
<-	routing_label		m
<-	circuit_identification_code		m
ISUP_body	ISUP_ACM		ISUP Body

Detailed Comment:	

The Structured Type ISUP_ACM has been defined. This Structured Type which contains all the fields of the ISUP ACM PDU except routing label and circuit identification code, can be used in both ISUP PDU definition and SIP message definition.

The SIP Response message that carries an ISUP ACM PDU is defined as:

PDU Type Definition	
PDU Name:	SIP_RESPONSE_ACM
Group:	
PCO Type:	SIP_TRANSPORT
Encoding Rule Name:	
Encoding Variation:	
Comments:	SIP RESPONSE including ISUP ACM

Field Name	Field Type	Type Encoding	Comments
status_line	Status_Line_Type		
headers_str	IA5String		Large String including all headers fields
headers	Message_header_type		
body	IA5String		
ISUP_body	ISUP_ACM		ISUP body with ACM PDU

Detailed Comment:	

The Structured Type ISUP_ACM is defined bellow (first field is the message type):

Structured Type Definition	
Type Name:	ISUP_ACM
Group:	
Encoding Variation:	
Comments:	

Element Name	Type Definition	Type Encoding	Comments
MType	message_type		m
BCI	backward_call_indicators		m
opt_part_ptr	pointer		m
OBCI	optional_backward_call_indicators		o
CRef	call_reference		o @
Cause	cause_indicators		o
UUInd	user_to_user_indicators		o
UUInf	user_to_user_information		o
ATP	access_transport		o
GenNot	generic_notification_indicator		o 1.
TMU	transmission_medium_used		o
EchoInf	echo_control_information		o
ADInf	access_delivery_information		o
RnNb	redirection_number		o
ParCmp	parameter_compatibility_information		o
CDInf	call_diversion_information		o
NtwFac	network_specific_facility		o @
RemOp	remote_operations		o @
ServAct	service_activation		o @
RnNbRes	redirection_number_restriction		o
ConTrInd	conference_treatment_indicators		o
UIDAcInd	UID_action_indicators		o
EndOP	end_of_optional_parameters_indicator		o

Detailed Comment:	

2. Examples of Test Cases

2.1 SIP to ISUP Test Case

2.1.1 Example of RFC requirement

From rfc 3272 ‘4.1 Originator’:

```
…
In the case of calls originating in the PSTN (see Figure 3 and Figure
5), the originating gateway takes the necessary steps to preserve the
ISUP information by encapsulating it in the SIP request it creates.
…
```

From rfc3398 ‘7.2.1.1 INVITE to IAM procedures’:

```
…
For example, if an INVITE arrives at a gateway with an encapsulated
IAM with a CPN field indicating the telephone number +12025332699,
but the Request-URI of the INVITE indicates 'tel:+15105550110', the
gateway MUST use the telephone number in the Request-URI, rather than
the one in the encapsulated IAM, when creating the IAM that the
gateway will send to the PSTN.
…
```

2.2 ISUP to SIP Test Case

From rfc 3272 ‘4.1 Originator’:

```
…
In the case of calls originating in the PSTN (see Figure 3 and Figure
5), the originating gateway takes the necessary steps to preserve the
ISUP information by encapsulating it in the SIP request it creates.
…
```

From rfc3398 ‘8.2.1.1 IAM to INVITE procedures’:

```
…
When an IAM arrives at a PSTN-SIP gateway, a SIP INVITE message MUST
be created for transmission to the SIP network.  This section details
the process by which a gateway populates the fields of the INVITE
based on parameters found within the IAM.

The context of the call setup request read by the gateway in the IAM
will be mapped primarily to two URIs in the INVITE, one representing
the originator of the session and the other its destination.  The
former will always appear in the From header (after it has been
converted from ISUP format by the procedure described in Section 12),
and the latter is almost always used for both the To header and the
Request-URI.
…
```