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Technical Specification

Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; Technology Mapping; Part 1: Implementation of TIPHON architecture using SIP

Reference

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ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

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Foreword

This Technical Specification (TS) has been produced by ETSI Project Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON).

The present document is part 1 of a multi-part deliverable covering implementation, of TIPHON architecture using the SIP protocol, as identified below:

Part 1: "Implementation of TIPHON architecture using SIP";

Part 2: "Implementation Profile for SIP".

1 Scope

The present document describes how the SIP protocol [9] completed with correlated protocols like SDP [11], HTTP [12] can be a candidate for TIPHON release 4 according to guidelines given in TS 101 315 Release 4 (see bibliography) and TS 101 315 Release 3 [2].

The SIP profile is derived from the examination of the following TIPHON Release 4 documents:

- the TIPHON baseline architecture described in TS 101 314 [1];
- the capabilities service required by TS 101 878 (see bibliography);
- the Meta-protocol as defined in multi part document TS 101 882-1 [5], TS 101 882-2 [6], TS 101 882-3 (see bibliography), TS 101 882-4 (see bibliography) and TS 101 882-5 [7];
- the end-to-end Quality of Service defined in TS 102 024-3 [3];
- the Security service defined in TS 102 165-1 [4].

The mapping of Meta-Protocol to SIP is limited to the following parts, while other parts are not available yet like supplementary services:

- Registration Meta-Protocol [6];
- Simple Call Meta-Protocol (TS 101 882-3 - see bibliography);
- Media Control Meta-Protocol (TS 101 882-4 - see bibliography);
- the end-to-end Quality of Service defined in TS 102 024-3 [3];
- IETF RFC 3261: "SIP: Session Initiation Protocol" [9];
- IETF RFC 2327: "SDP: Session Description Protocol" [11];
- IETF RFC 2616: "Hypertext Transfer Protocol - HTTP/1.1" [12];
- IETF RFC 2617: "HTTP Authentication: Basic and Digest Authentication" (see bibliography);

Furthermore the following documents have been consulted for information:

- TS 124 229: "IP Multimedia Call Control Protocol based on SIP and SDP" (see bibliograpy);
- TS 124 228: "Signalling flows for the IP multimedia call control based on SIP and SDP" [8].

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 and/or edition number or version number) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

[1] ETSI TS 101 314: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; Abstract Architecture and Reference Points Definition; Network Architecture and Reference Points".

[2] ETSI TS 101 315: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; Functional entities, information flow and reference point definitions; Guidelines for application of TIPHON functional architecture to inter-domain services".

[3] ETSI TS 102 024-3: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; End-to-end Quality of Service in TIPHON Systems; Part 3: Signalling and Control of end-to-end Quality of Service".

[4] ETSI TS 102 165-1: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; Protocol Framework Definition; Methods and Protocols for Security; Part 1: Threat Analysis".

[5] ETSI TS 101 882-1: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; Protocol Framework Definition; Part 1: Meta-protocol design rules, development method, and mapping guideline".

[6] ETSI TS 101 882-2: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; Protocol Framework Definition; Part 2: Registration and Service Attachment service meta-protocol definition.".

[7] ETSI TS 101 882-5: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; Protocol Framework Definition; Part 5: Transport control service meta-protocol definition;".

[8] ETSI TS 124 228: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Signalling flows for the IP multimedia call control based on SIP and SDP; Stage 3 (3GPP TS 24.228 version 5.3.0 Release 5)".

[9] IETF RFC 3261: "SIP: Session Initiation Protocol".

[10] IETF RFC 3264: "An Offer/Answer Model with Session Description Protocol (SDP)".

[11] IETF RFC 2327: "SDP: Session Description Protocol".

[12] IETF RFC 2616: "Hypertext Transfer Protocol - HTTP/1.1".

[13] IETF RFC 2617: "HTTP Authentication: Basic and Digest Access Authentication".

[14] IETF RFC 1890: "RTP Profile for Audio and Video Conferences with Minimal Control".

[15] IETF RFC 1889: "RTP: A Transport Protocol for Real-Time Applications".

[16] IETF RFC 2806: "URLs for Telephone Calls".

[17] IETF RFC 2748: "The COPS (Common Open Policy Service) Protocol".

[18] IETF RFC 2326: "Real Time Streaming Protocol (RTSP)".

[19] IETF RFC 3525: "Gateway Control Protocol Version 1".

[20] IETF RFC 3265: "Session Initiation Protocol (SIP)-Specific Event Notification".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in TS 101 314 [1] and TS 101 878 (see bibliography) apply.

3.2 Abbreviations

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

B2BUA	Back-to-Back User Agent
BC	Bearer Control
CC	Call Control
COPS	Common Open Policy Service
FG	Functional Group
ICF	Inter-Connect Function
IP	Internet Protocol
MC	Media Control
NFG	Network Functional Group
PCM	Pulse Code Modulation
PDP	Policy Decision Point
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RPC	Remote Procedure Call
SAP	Service Access Point
SC	Service Control
SDP	Session Description Protocol
SpoA	Service point of Attachment
TE	Terminal Equipment
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
URI	Uniform Resource Identifier

4 SIP environment overview

4.1 Introduction

The purpose of the present document is not to describe how to implement SIP protocol but how TIPHON protocol can be represented in a SIP environment. For example parameter mandatory in SIP but without equivalence in TIPHON information elements are not documented. Mandatory behaviours in SIP that do not correspond to any TIPHON behaviours are not documented either.

The aim is to identify gap in TIPHON to SIP direction between both protocols. Informative suggestions to fill those gaps could be given in conclusion.

4.2 SIP protocol

SIP is a relatively new technology (1995) developed for remote control, establishment and tear-down of multimedia sessions. The origins of SIP are in the academic and IETF community and assumed in its first incarnation a public internet although with the interest shown by 3GPP the application to a managed network that uses IP has become ascendant. SIP is based upon the communication model of HTTP and therefore is broadly viewed as a request-response protocol. In relation to other well known protocols SIP has close cousins in Remote Procedure Call (RPC) and in the ITU-T ROSE protocol.

According to RFC 3261 [9], SIP is an application-layer-control protocol to manage multimedia session. But, "SIP is not a vertically integrated communications system", and will need other IETF protocols to build a complete multimedia architecture (e.g.: RTP RFC 1889 [15], RTSP RFC 2326 [18], MEGACO RFC 3525 [19], SDP RFC 2327 [11]).

The choice of the protocol for the session description is opened in SIP and appears in SIP only as a parameter value (Content-Type). The media type descriptions that can be included in the body of a SIP message are Internet Media Types as in HTPP/1.1. However, in this profile only Session Description Protocol (SDP) defined in RFC 2327 [11] has been considered. SIP reuses also the authentication mechanism defined in HTTP.

The SIP technology has been considered through the following standards:

- "SIP: Session Initiation Protocol" - RFC 3261 [9].
- "SDP: Session Description Protocol" - RFC 2327 [11].
- "RTP Profile for Audio and Video Conferences with Minimal Control" - RFC 1890 [14].
- "Hypertext Transfer Protocol - HTTP/1.1" - RFC 2616 [12].

SIP does not define services. Rather, SIP provides primitives that can be used to implement different services. For example, SIP can locate a user and deliver an opaque object to his current location. If this primitive is used to deliver a session description written in SDP, for instance, the endpoints can agree on the parameters of a session. If the same primitive is used to deliver a photo of the caller as well as the session description, a "caller ID" service can be easily implemented. As this example shows, a single primitive is typically used to provide several different services.

SIP does not offer conference control services such as floor control or voting and does not prescribe how a conference is to be managed. SIP can be used to initiate a session that uses some other conference control protocol. Since SIP messages and the sessions they establish can pass through entirely different networks, SIP cannot, and does not, provide any kind of network resource reservation capabilities.

The nature of the services provided make security particularly important. To that end, SIP provides a suite of security services, which include denial-of-service prevention, authentication (both user to user and proxy to user), integrity protection, and encryption and privacy services.

SIP works with both IPv4 and IPv6.

4.2.1 SIP signalling, methods and responses

4.2.1.1 SIP signalling

The SIP protocol client/server machine is very simple: Request is sent and the requestor (client) waits for a response. The request contains the method and who the method is aimed at, the response contains the status code that informs the requestor of how the server has dealt with the request.

4.2.1.2 Methods and responses

There are 6 core methods in SIP and these are the basis of the protocol:

- INVITE - starts a session (and modifies it if used as a re-invite).
- ACK - confirms the invite.
- BYE - terminates a sessions.

- CANCEL - cancel an invite.
- OPTIONS - Querying capability.
- REGISTER - binds a user's address (SIP name) to a network address (IP address).

4.2.2 SIP protocol components

The protocol of SIP is enabled by assigning particular functions to a set of protocol components. A particular SIP device will contain 1 or more of these components.

- User Agent Client (UAC).
- User Agent Server (UAS).
- Redirect server.
- Proxy server.
- Registrar.

The UAC and UAS exist in a normal terminal device and are termed jointly the User Agent.

The proxy server arises from breaking the assumption that the UACs know the UASs that they want to communicate with. In anything but the smallest of networks this assumption is inevitably broken so a network resident proxy to the UA exists to facilitate routing.

- Proxy servers can be configured to perform inter-domain call establishment.
- The registrar server is a special server that attends to REGISTER methods. In most cases the registrar and proxy server will be co-located.

4.3 SDP

SDP is a session description protocol in text format language. It is used in SIP to define a simple offer/answer model to a describe unicast session. Mapping in the present document has been based on RFC 2327 [11] overloaded with RFC 3264 [10].

4.4 HTTP/1.1

Hypertext Transfer protocol provides a scheme description for authentication.

According to RFC 3261 [9], chapter 22, only the "Digest" authentication mechanism described in RFC 2617 [13] overload by RFC 3261 [9] has to be considered.

5 Implementation of TIPHON functional architecture using SIP

5.1 Introduction

5.2 SIP functional architecture

The SIP Architecture has the following functional elements, as defined in [9].

User Agent (UA): The user agent is the functional entity that may initiate or respond to a SIP request.

In a TIPHON compliant system, the SIP User Agent (UA) shall provide the functionality of the terminal functional group. The terminal functional group performs the roles of the terminal registration functional group, originating terminal functional group and the terminating terminal functional group. The reference points S1, SC1 and N1 are regarded as internal to the TE.

Back-to-Back User Agent (B2BUA): B2BUA is a logical entity that receives a request and processes it as a User Agent Server (UAS). In order to determine how a request should be answered, it acts as a User Agent Client (UAC) and generates requests. Unlike a proxy server (stateless), it maintains a dialogue state, and must participate in all requests sent on the dialogues it has established. TIPHON recommends the use of a B2BUA, as network functional groupings involved in providing a service.

Proxy server: A proxy server acts as both the client and server: It receives a request from an entity, and initiates a request on behalf of the requesting entity, hence acting as a server for the requesting entity. A proxy server can be stateless (forgets about the state of a particular session) or statefull (keeps track of the state of the session it is involved in).

Redirect server: A redirect server receives requests from an entity, and returns the contact address of the destination to the resquesting entity.

Registrar: The registrar processes registration requests; as a minimum this involves updating the users contact list and responding to the originator of the request. Typically a registrar is co-located with either the proxy or the redirect server, and may be adapted to perform location-based services.

SIP gateway: A SIP gateway acts as an interworking medium between the PSTN and a SIP network. It provides an interworking between SIP and PSTN call control protocols, such as ISUP, as well as interworking between the TDM and IP media flows. A SIP gateway can be decomposed into a gateway controller (taking cate of the call control protocol conversion) and a media gateway (taking cate of the TDM to IP media conversion).

Figure 1 shows how the SIP functional elements map onto the functional layers in the IP Telephony Application plane.

Figure 1: Void

The UA maps to Service, Service Control (SC), Call Control (CC), and Bearer Control (BC) layers.

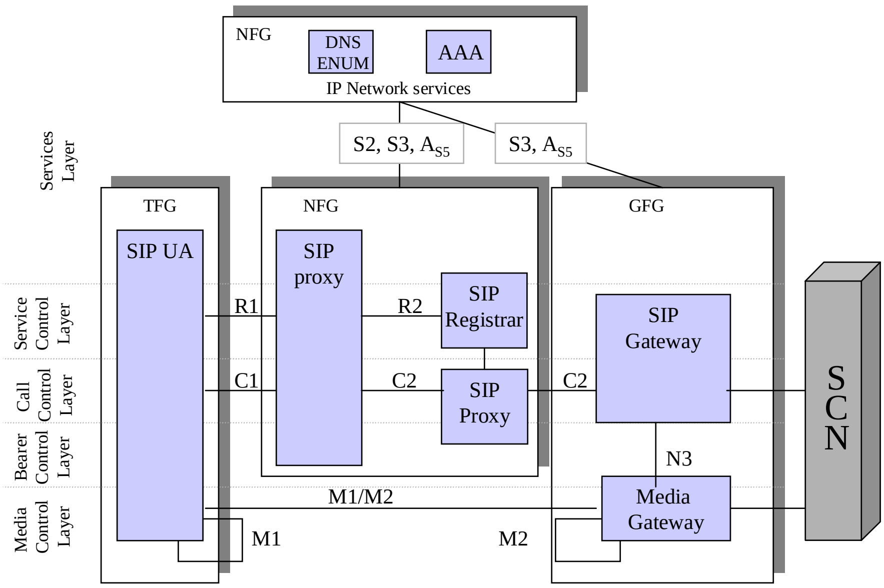
The statefull and stateless proxy maps to the TIPHON service control, call control and bearer layers.

The SIP gateway covers all TIPHON layers.

The redirect server works at TIPHON service control and call control layers.

The registrar works at TIPHON Service and Service Control layer.

Figure 2 shows the SIP entities and how they map to the functional layers and the Functional Groups (FG) defined in TS 101 314 [1].



NOTE 1: All entities in an IP network "normally" use the DNS service. In the context of the present document only relations to the DNS with ENUM extensions are shown.

NOTE 2: The gateway shown is a decomposed gateway (combination of a gateway controller and a media gateway).

Figure 2: SIP Architecture mapped to the TIPHON Functional layers and functional groups

The SIP proxy, SIP gateway and the SIP Registrar shall provide the functionality required in the Network Functional Group (NFG). Reference point S2, S3 and A_{S5} are between the Network Functional Group and other IP Network services e.g. DNS. The Network Functional Group may play the roles of an originating Network Functional Group, an intermediate Network Functional Group or a terminating Network Functional Group.

NOTE: The Network Functional Group may include Media Control Functional Entities, e.g. for giving announcements, mixing media streams etc. This is, however, out of scope of the present document.

The present document describes the mapping of functional architecture TS 101 314 [1], as well as the context, behaviour and procedures TS 101 882-1 [5] that the SIP and SDP protocols must adhere to, to be TIPHON compliant. In TIPHON Release 4, SIP is mapped to reference points R1, R2, C1, C2, where R1 and R2 refer to the registration reference points, whereas C1 and C2 refer to call & bearer control reference points. The R and C reference points will be dealt with separately in the present document, because of the different nature of services they provide.

6 Registration service

6.1 Introduction

According to the meta-protocol defined in TS 101 882-2 [6] and functional the description defined in TS 101 315 [2], the purpose of the TIPHON registration service includes the authentication and authorization of a subscriber (user/registrant) to access a service.

The basic registration mechanism can be described as follows:

1) User registration: The user registers for the service and shows entitlement for the service used.

2) Service preparation: The registrar selects a service node at which the user shall use the service and informs the service node that the user is entitled to use the service.

3) Service attachment: The user (terminal) attaches to the service node and the service can be delivered.

Two registration scenarios shall be supported:

- the "User at home" scenario;
- the "Roaming user" scenario.

Registration in SIP is part of a location service. With the REGISTER message a UA informs location server how it could be contacted. This functionality is a bit far from Registration service as defined in TIPHON. However, the REGISTER message contains a Proxy-Authorization header field that allows an authentication and authorization mechanism. This mechanism can be explicitly requested by the Service point of Attachment with a Proxy-Authenticate header in a 407 (Proxy Authentication Required) Response. This field will be set by the UA and analysed by the Proxy SpoA. RFE1 and RFE2 have to be the same SIP entity.

Consequently, there is not always a one to one mapping between TIPHON registration information flow sequence and SIP registration signalling. For example the UA sends one or two REGISTER messages depending if the UA is waiting for 401, 407 responses before setting Proxy-Authorization header. The REGISTER message will cover both information flows Registration_req and Authorize_r. In case of "User at home" RFE2 and RFE3 can be considered as a SIP outbound proxy and the REGISTER message is mapped with Registration_req and Authorize_req. In case of "Roaming user" the REGISTER message will be forwarded by proxies that behave as Originating NFG and intermediate FG to RFE2/RFE3. The initial REGISTER message shall contain information useful in both Registration_req and Authorize_req and will be set by the UA.

Additionally, in case of "Roaming user", an intermediate proxy between RFE1 and RFE2 may require a SIP registration from RFE1 before any TIPHON registration. This is implementation dependant. In pure IP environment RFE1 can address directly its registration to RFE2 or has to go through an intermediate proxy. In both case it will have to know the IP address, port and transport protocol of its home network. This can be done statically. The UA will have to know also its current domain name address to set at its contact address.

De-Registration and Registration in SIP are covered by the same protocol message REGISTER.

According to RFC 3261 [9], chapter 22 "Basic" authentication is not allowed in SIP, only "Digest" authentication mechanism described in RFC 2617 [13] overload by RFC 3261 [9] can be used. Authentication parameters value is given in the following table for a "Digest" mode. Authentication parameters are given in unsuccessful message in SIP while they are expected in successful message in TIPHON. This makes some distortion in the mapping.

Deregistration in SIP uses the same REGISTER message with the expire parameter set to zero on the contact to remove or a contact list set to * (meaning all contact) and an Expires header field set to 0. The registrar cannot ask to the user for deregistration.

A SIP registration signalling flow could be:

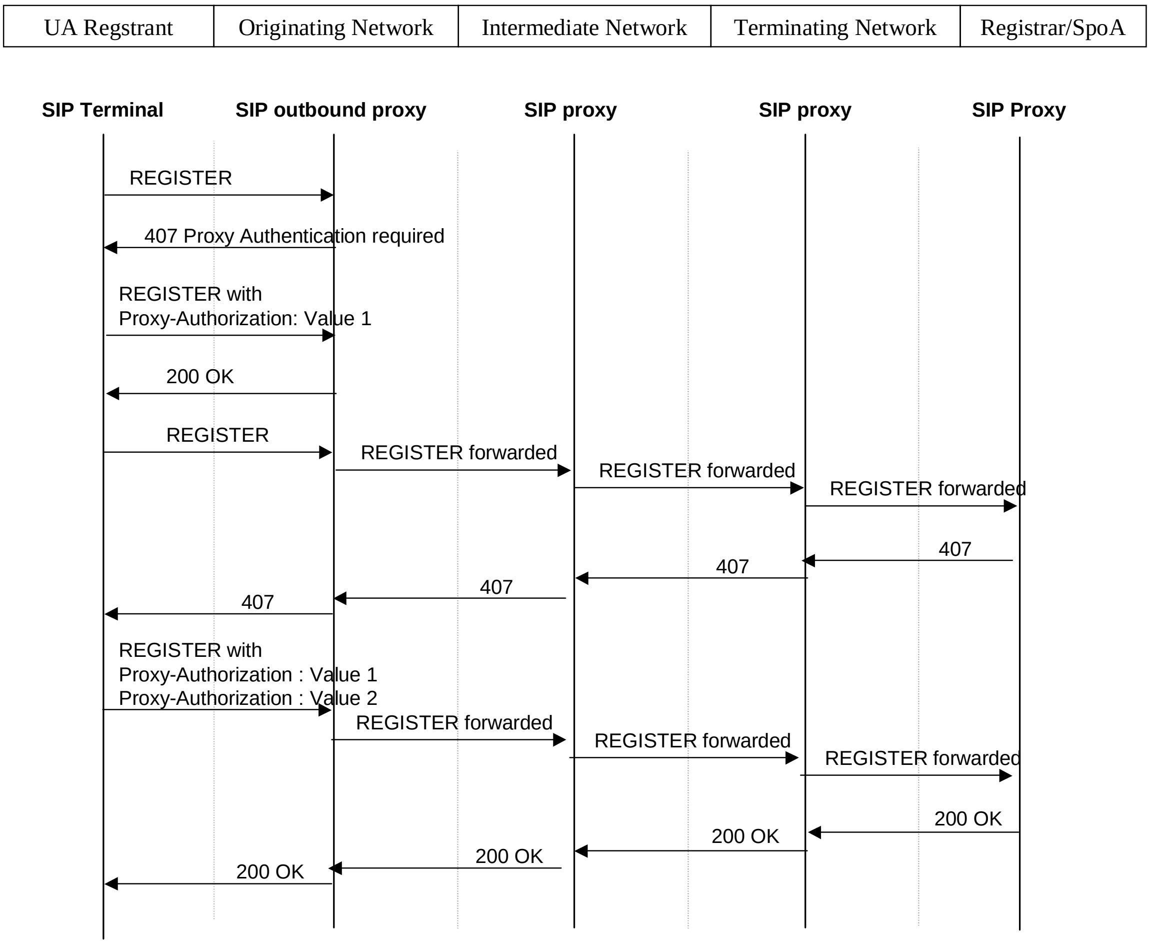


Figure 3: The SIP registration example

6.2 Registration functional entities mapping

Table 1: Mapping of SIP entities to TIPHON Registration Functional entities

TIPHON functional element		SIP entities	
Identity	**TIPHON Description**	**User at Home**	**Roaming User**
RFE1	Registrant, the logical entity being registered	UA	UA
RFE2	Registrar, holder of user profile of the registrant	Registrar	Registrar
RFE3	Serving Service Provider point of Attachment (SpoA)	Outbound Proxy	Home Proxy
RFE4	Previous SpoA	Proxy	Proxy

6.3 Registration Messages Mapping

Table 2 shows the mapping of TIPHON Registration meta-protocol information flows to SIP messages.

Responses to these requests, when a SIP message corresponds, are done with SIP responses described in RFC 3261 [9], chapters 7.2 and 20. A 2XX answer corresponds to a positive answer while a 4XX-6XX answers to a reject.

Table 2: Mapping of SIP messages to TIPHON Registration MPMUs

TIPHON message	RelationShip ID	SIP messages
Register	RFE1<->RFE2	REGISTER/SIP Response
DeRegister	RFE1<->RFE2	REGISTER with contact header field set to * and Expires header field set to 0/SIP Response
Authorize	RFE2<->RFE3	None
Detach	RFE2<->RFE3	None
Attach	RFE3<->RFE1	Proxy-Authorization header field in SIP message concerned by the service invocated
Detach	RFE2<->RFE4	None

6.4 Registration information flow Mapping

6.4.1 Relationship ra (RFE1/RFE2)

Table 3: Mapping of SIP to Register request from RFE1 to RFE2

Register *request/indication*					
TIPHON				**SIP**	
Information element		**Status**	**Value**	**Mapping**	**Notes**
TIPHON-reg-id		M	Any	URI in the TO header (address-of-record) and username parameter in Proxy-Authorization header	TO and FROM SIP header can differ while for roaming user scenario a third party Registration is needed
RegistrationMode		M	Initial registration Location update	None	There is no distinction between Initial registration and updating Contact in SIP from data point of view. Only behaviour requesting the authorization (401, 407 Responses) can reflect a first registration.
Location (of Registrant)		M		Contact header set to the current location of the registrant	
	protocolID			Fixed value 'sip' used in addr-spec	
	nameorAddress			Name-addr or addr-spec of Contact	
	port			Port of hostport of addr-spec used in Contact	
ServiceName		M	TIPHON Simple Call…	Could be set as part of realm value in Proxy-Authorization header	

Table 4: Mapping of SIP to Register response from RFE2 to RFE1

Register *response/confirmation*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
TIPHON-reg-id	M	Any	URI in the TO header (address-of-record)	
ServiceName	O (see note 2)		None	
Result	M	Registration successful, Registration-Id invalid, Service unavailable	Status-Code	200 OK/407 Proxy Authentication require?, 406 Not Acceptable, 503 Service Unavailable
ServiceProviderName	O (see note 1)	Any	None	
ClientAuhorizationToken	O (see note 1)	Any	Opaque parameter of Proxy-Authenticate header	(see note 1) According to this note and note 3, there is no mapping in flow sequence
NOTE 1: Provided if Result='Registration successful'. NOTE 2: Provided if Result='Service unavailable'. NOTE 3: This value will be present only 407 reject response.				

Table 5: Mapping of SIP to DeRegister request from RFE1 to RFE2

DeRegister *request/indication*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
TIPHON-reg-id	M	Any	URI in the TO header (address-of-record) and username parameter in Proxy-Authorization header	
ServiceName	M	TIPHON Simple Call…	Could be set as part of realm value in Proxy-Authorization header	

Table 6: Mapping of SIP to DeRegister response from RFE2 to RFE1

DeRegistration *response/confirmation*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
TIPHON-reg-id	M	Any	URI in the TO header (address-of-record)	
Result	M	Deregistration successful, Registration-Id invalid	Status-Code	200 OK, 406 Not Acceptable/407 Proxy Authentication require?

6.4.2 Relationship rb (RFE2/RFE3)

Even if no SIP signalling flow is defined between RFE2 and RFE3 a pseudo mapping has been tried with the data contains in the REGISTER message received by RFE2.

Table 7: Mapping of SIP to Authorize request from RFE2 to RFE3

Authorize *request/indication*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Registrar-id	M	Any	Domain set in uri parameter of Proxy-Authorization header	
TIPHON-reg-id	M	Any	URI in the TO header (address-of-record) and username parameter in Proxy-Authorization header	
ServiceName	M	TIPHON Simple Call…	Could be set as part of realm value in Proxy-Authorization header	

Table 8: Mapping of SIP to Authorize response from RFE3 to RFE2

Authorize *response/confirmation*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Registrar-id	M	Any	Domain set in domain parameter of Proxy-Authenticate header	note 1
TIPHON-reg-id	M	Any	URI in the TO header (address-of-record)	
ClientAuthorizationToken	O (note 2)	Any	Opaque parameter of Proxy-Authenticate header	notes 1 and 2 create a mismatch in the flow
Result	M	ServiceAuthorized to Client, ResourceNot available	Status-Code	200 OK, 503 Service Unavailable
NOTE 1: This value will be present only 407 reject response and will have to be picked up before a successful answer. NOTE 2: This information element shall be provided if the value of Result is 'OK'.				

Table 9: Mapping of SIP to Detach request from RFE2 to RFE3

Detach *request/indication*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Registrar-id	M	Any	Domain set in uri parameter of Proxy-Authorization header	
TIPHON-reg-id	M	Any	URI in the TO header (address-of-record) and username parameter in Proxy-Authorization header	
ServiceName	M	TIPHON Simple Call…	Could be set as part of realm value in Proxy-Authorization header	

Table 10: Mapping of SIP to Detach response from RFE3 to RFE2

Detach *response/confirmation*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Registrar-id	M	Any	Domain set in domain parameter of Proxy-Authenticate header	(see note 1)
TIPHON-reg-id	M	Any	URI in the TO header (address-of-record)	
Result	M	Service detachment successful Identity not recognized	Status-Code	200 OK, 404 Not Found/407 Proxy Authentication require?
NOTE 1: This value will be present only 407 reject response. NOTE 2: This information element shall be provided if the value of Result is 'OK'.				

6.4.3 Relationship rc (RFE1/RFE3)

The data are mapping only with the header field SIP Proxy-Authorization included in the SIP message correlated to the service invocated.

Table 11: Mapping of SIP to Attach request from RFE2 to RFE3

Attach *request/indication*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Registrar-id	M	Any	Domain set in uri parameter of Proxy-Authorization header	
TIPHON-reg-id	M	Any	username parameter in Proxy-Authorization header	
ServiceName	M	TIPHON Simple Call…	Could be set as part of realm value in Proxy-Authorization header	
AuthorizationToken	M	Any	Opaque parameter of Proxy- Authorization header	

Table 12: Mapping of SIP to Attach response from RFE3 to RFE2

Attach *response/confirmation*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Registrar-id	M	Any	None	
TIPHON-reg-id	M	Any	None	Proxy-Authenticate header is not sent in response when a valid Proxy-Authorization has been sent in the request
Result	M	Service attachment successful Identity, not recognized, Authorization expired	Status-Code	200 OK, 407 Proxy Authentication require, None

6.4.4 Relationship rd (RFE2/RFE4)

No mapping can be done.

6.5 Registration action Mapping

Table 13: Mapping of SIP to Registration action at RFE1

Actions at RFE1	
TIPHON Action number	**SIP behaviour**
101	Preparation and Sending of first REGISTER message without Proxy-Authorization.
102 for initial Registration	Pick up Proxy-Authenticate header in the 407 answer.
102 for location update	Wait for 200 OK response containing the new contact list.
103	Sending of an additional REGISTER message with a valid Proxy-Authorization header.
104	Wait for 200 OK response, save this Proxy-Authorization header for future use in request.
105	Preparation and Sending of a REGISTER message with Proxy-Authorization header and new contact header to update the location
106	None
107	Preparation and Sending of a REGISTER message with Proxy-Authorization header, contact header field set to * and Expires header field set to 0
108	Wait for 200 OK response

Table 14: Mapping of SIP to Registration action at RFE2/RFE3

Actions at RFE2/RFE3	
TIPHON Action number	**SIP behaviour**
201/202/203/204 301/302/303/304 for initial Registration	Preparation and Sending of a 407 Answer with a Proxy-Authenticate header.
201/202/203/204 for location update	Preparation and Sending of a 200 OK answer with updated contact list.
205/305/306	Answering with a 4xx-6xx response
206	None
207	Updating of the contact list
208/209/307	Updating of the contact list
210/308	Answering with a 200 OK response without any contact
211	Answering with a 4xx-6xx response

Table 15: Mapping of SIP to Registration action at RFE4

Actions at RFE4	
TIPHON Action number	**SIP behaviour**
401	None
402	None

6.6 Conclusion

The SIP registration service does not cover completely the TIPHON meta-protocol intention. Additional IETF extension to the protocol should be considered.

Concerning DeRegistration initiated by RFE2, RFC 3265 [20] has been already studied. RFC 3265 [20] allows to manage events for registrations in SIP. RFC 3265 [20] will allow exchange (NOTIFY) between Registrar RFE2 and Service providers RFE3 and RFE4. But this implies that first RFE3 and RFE4 subscribe to RFE2, which is not described in the meta-protocol and additionally RFE3 and RFE4 will receive yet only information on the registration state for a particular address-of-record ("init", "active", "terminated"). The Authorization mechanism with the registrant still has to be managed in the registrar. This mechanism is used in 3GPP to ask with a NOTIFY sent to the registrant a deregistration (RFE1 is the subscriber).

7 Simple call application

7.1 Introduction

The intentions with this clause is to describe the simple call application defined in the Meta-protocol in TS 101 882-3 (see bibliography) using procedures defined in SIP (RFC 3261 [9]) and map those procedures to the architecture of TS 101 314 [1]. RFC 3264 [10] has been considered in this process concerning SDP (RFC 2327 [11]).

Two scenarios shall be supported:

- the "user at home" scenario; and
- the "roaming user" scenario.

NOTE: For details about the two scenarios (including some examples) see the TS 101 315 (see bibliography).

The simple call application includes following services:

1) A calling User establishes a call via its home network with a called User.
2) An authorization mechanism based on the result of the registration is proceeded.
3) This call can be released either by the calling User, the called User or the network.
4) The media path is reserved and connected while the call is establishing.

The ringing tone is local in TIPHON and not a bearer on a media channel.

It is implicit in TIPHON that the release message will go through the same node as the call Setup where resources have been reserved. To guarantee this, record-route and route procedures in SIP will have to be used while the call is established by the proxy in charge of the media.

In case of roaming scenario, several proxies can act between the Calling User and its Home proxy (covered by CFE1, CFE2, CFE3, CFE4, CFE5) but have no correspondence in simple call functional entities.

The UA will have to include in its INVITE a proxy-authorization header field set to the parameter value received during the registration.

The SIP simple call signalling flows could be as in following figures.

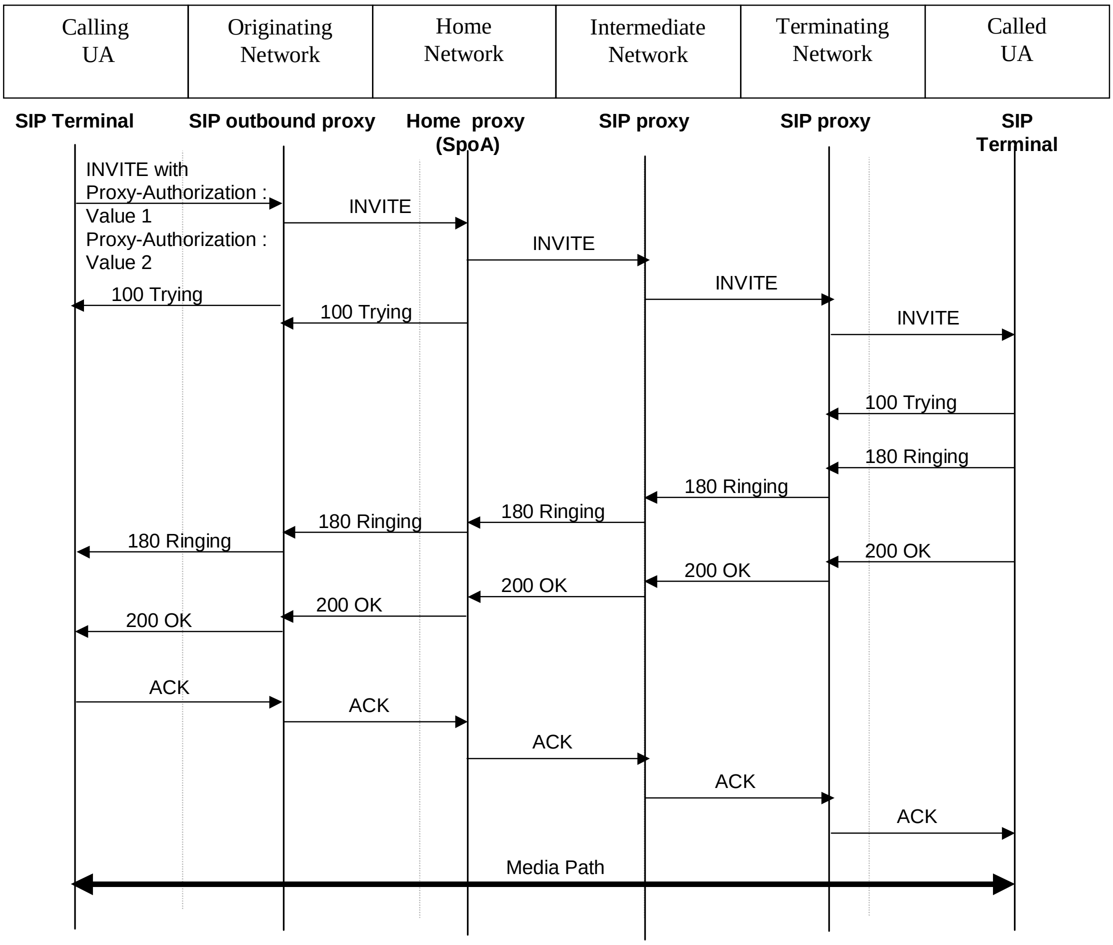


Figure 4: SIP successful simple call

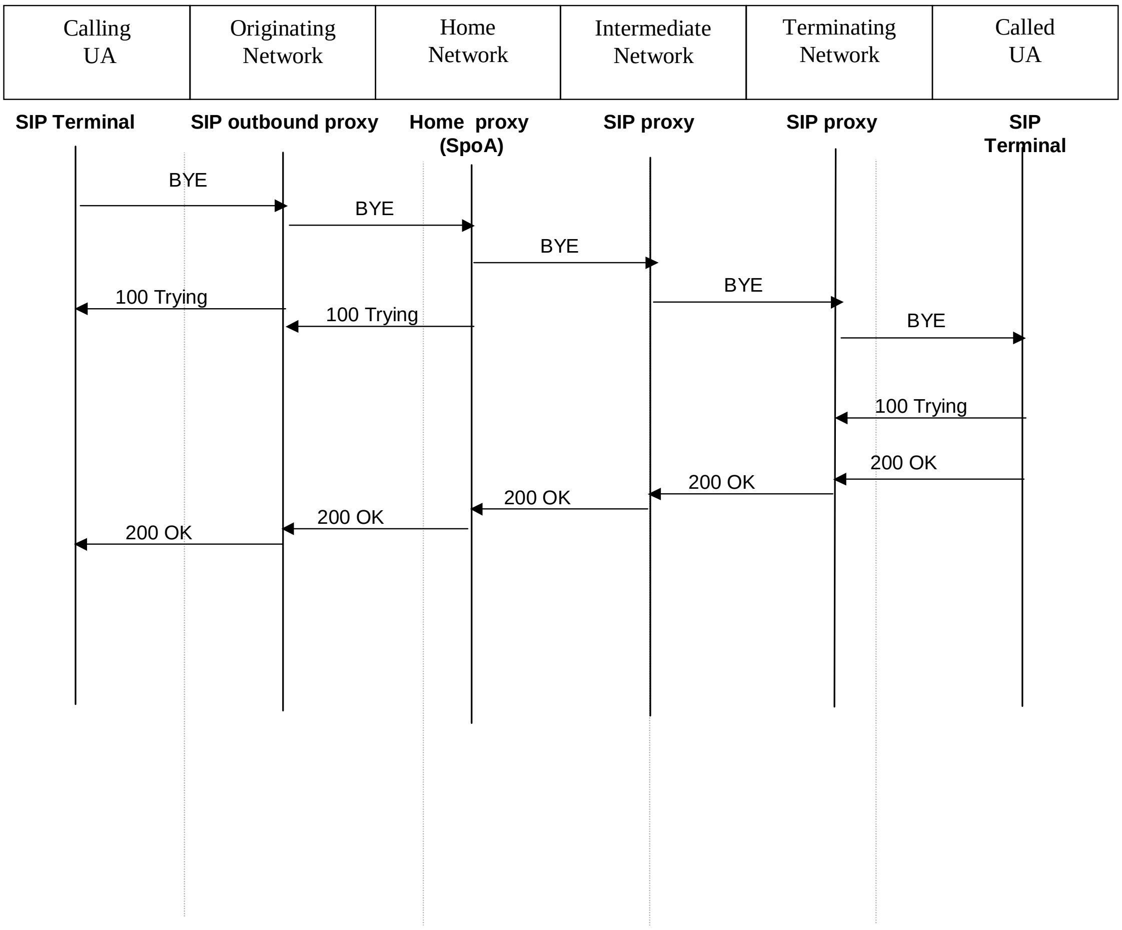


Figure 5: SIP call release while a call has been established

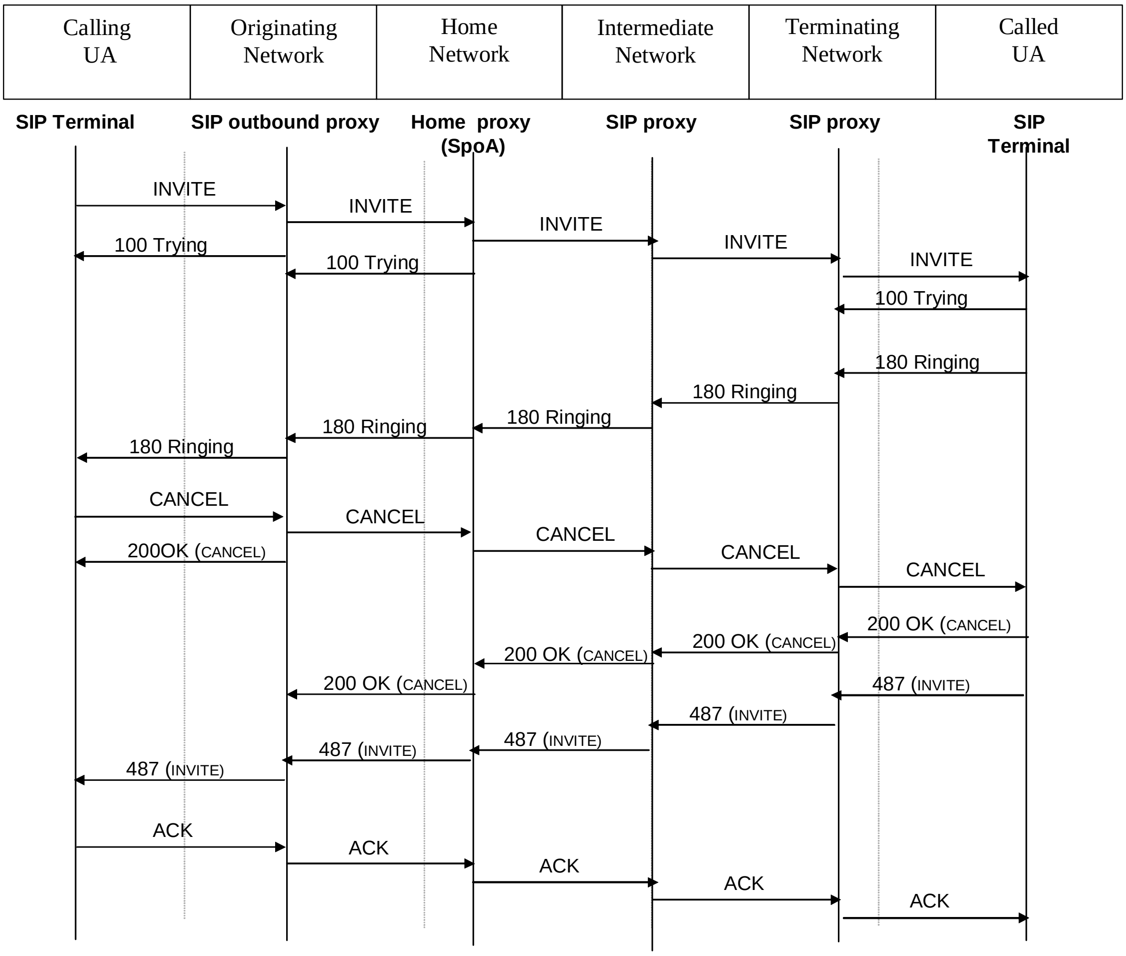


Figure 6: SIP call release while a call is still not established

7.2 Simple call functional entities mapping

Table 16: Mapping of SIP entities to TIPHON simple call functional entities

TIPHON functional element		SIP entities	
Identity	**TIPHON Description**	**User at Home**	**Roaming User**
Calling User	The application at the calling user''s terminal which instigates the service request	UA Client	UA Client
$CFE1_{OUA}$	The originating user service agent in the calling user''s terminal that instigates the service request	Home stateful Proxy in the same current domain of the UA (Server Part)	Home stateful Proxy (Server Part)
$CFE2_{PE}$	The serving network policy control function associated with the calling user''s service provider	Home stateful Proxy in the same current domain of the UA (Registrar part)	Home stateful Proxy (Registrar part)
$CFE3_{OCC}$	The originating call coordination function that is responsible for establishing the call on behalf of the calling user	Home stateful Proxy in the same current domain of the UA (Client Part)	Home stateful Proxy (Client Part)
$CFE4_{OR}$	The originating call routing function, providing routing information and number/address translations	DNS server/Location Server	SIP routing rules, DNS server, Location Server
$CFE5_{OT}$	The originating transport coordination function serving the calling user	RSVP	RSVP
$CFE6_{ICC}$	An intervening call control coordination function. This CFE is responsible for establishing the call via the intervening domain	Intermediate stateful Proxy	Intermediate stateful Proxy
$CFE7_{IR}$	An intervening routing function	DNS server/Location Server?	SIP routing rules, DNS server, Location Server
$CFE8_{IT}$	An intervening transport coordination function	RSVP	RSVP
$CFE9_{TCC}$	The destination call coordination function that is responsible for establishing the requested call on behalf of the called user	stateful Proxy of called Domain	stateful Proxy of called Domain
$CFE10_{TT}$	The destination transport coordination function serving the called user	stateful Proxy of called Domain	stateful Proxy of called Domain
$CFE11_{TUA}$	The service agent that processes an incoming call to the called user	stateful Proxy of called Domain	stateful Proxy of called Domain
Called User	The application in the called user''s terminal at which the service request is terminated	UA Server	UA Server
NOTE: The mapping of SIP protocol entities to TIPHON functional entities shown in this table may not be the only mapping possible and it is recognized that alternative mappings may exist. The remainder of the present document assumes only the existence of the mapping shown in table 16.			

7.3 Simple call messages mapping

Table 17 shows the mapping of TIPHON simple call meta-protocol information flows to the SIP messages.

Responses to these requests, when a SIP message corresponds, are done with SIP responses described in RFC 3261 [9], chapters 7.2 and 20. A 2XX answer corresponds to a positive answer while a 4XX-6XX answers to a reject and 1XX to a provisional. To get an alerting at Calling user side, called user will have to send a provisional response 1XX that will be forwarded even if Alerting flow does not exist on called user side.

Table 17: Mapping of SIP messages to TIPHON simple call MPMUs

TIPHON message	RelationShip ID	SIP messages
TCC_OrigCallSetup	Calling User-<>CFE1	INVITE/SIP Response
TCC_CallRelease	Calling User<->CFE1	BYE/CANCEL/SIP Response
TCC_CallAlerting	Calling User<->CFE1	Provisional response like 183 Session Progress or 180 Ringing
ServingNwPolicy	CFE1<->CFE2	None internal
CallSetup	CFE1<->CFE3	None
CallRelease	CFE1->CFE3	None
CallAlerting	CFE1->CFE3	None
CallRoute	CFE3<->CFE4 CFE6<->CFE7	None internal or concern other protocol like DNS
TRMReserve	CFE3<->CFE5 CFE6<->CFE8 CFE9<->CFE10	Concern other protocol like RSVP
TRMConnect	CFE3<->CFE5 CFE6<->CFE8 CFE9<->CFE10	Concern other protocol like RSVP
TRMRelease	CFE3<->CFE5 CFE6<->CFE8 CFE9<->CFE10	Concern other protocol like RSVP
NWCallSetup	CFE3<->CFE6 CFE6<->CFE9	INVITE/SIP Response forwarded
NWCallRelease	CFE3<->CFE6 CFE6<->CFE9	BYE/CANCEL forwarded
NWCallAlerting	CFE3<->CFE6 CFE6<->CFE9	Provisional response like 183 Session Progress or 180 Ringing (except 100 that is local) forwarded
DestCallSetup	CFE9<->CFE11	None
CallRelease	CFE9<-> CFE11	None
TCC_DestCallSetup	CFE11<->Called User	INVITE/SIP Response
TCC_CallRelease	CFE11<->Called User	BYE/CANCEL/SIP Response

7.4 Simple call information flow mapping

Only Call information flows that have an equivalence in SIP have been mapped.

7.4.1 Relationship ra (CallingUser/CFE1)

Table 18: Mapping of SIP to TCC_OrigCallSetup request from CallingUser to CFE1

TCC_OrigCallSetup *request/indication*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Call Identifier	M (note 1)	Alphanumeric handle	Call-ID header	Complete call leg (Call-ID, To, From) should better identified the call
Called user ID	M	TIPHON user name	SIP-URI contained in To header	
Calling user ID restriction	M	Available/unavailable	None	Always set to available in SIP
Calling user ID	O (note 2)	TIPHON user name	SIP-URI contained in From header	Always present
Operator selection	O	OperatorSelection	None	
Service Offer Ticket	M	TicketType	Proxy-Authorization header	
registrantId	M	Visiblestring	username parameter in Proxy-Authorization header	
RegistrarId	M	Visiblestring	Domain set in uri parameter of Proxy-Authorization header	
serviceCredential	M	ServiceCredentialType		
serviceAppId	M	Visiblestring	Could be set as part of realm value in Proxy-Authorization header	
spoA	M	Visiblestring	Domain set in uri parameter of Proxy-Authorization header	In SIP SpoA is equivalent to the Registrar so its domain name can be reused
startTime	M	GeneralizedTime	None	
stopTime	M	GeneralizedTime	None	
cryptoDigest	O	Visiblestring	Opaque parameter of Proxy-Authorization header	note 3
cryptoDigest	O	Visiblestring	Opaque parameter of Proxy-Authorization header	note 3
QoSServiceClass	M	enumerated Value	SDP parameters	a=quality:<quality> SDP line could be used
TrafficDescriptor	M	TrafficDesc	SDP parameters	Derived to a=ptime:
peakFrameRate	M	Integer	None	
maxFrameLength	M	Integer	None	
Codec	M	List of possible codecs	SDP parameter encoding name in a=rtpmap: lines	note 4
Transcode count	M	Number of codec transcoding	Number of SDP lines starting with a=rtpmap: and with different encoding name set	note 4
Previous Domain Egress point	M	Network specific address	SDP parameter c= and port set in SDP parameter m=audio	A SDP parameter a= has to be set to sendrcv value. Because only one port can be set, one m=audio line shall be contained in the SDP offer.
NOTE 1: A temporary Call Identifier value may be used in the call setup request. NOTE 2: Shall be present if 'Calling User ID restriction' information element is set to value 'available'. NOTE 3: One at least has to be set. NOTE 4: For a RTP/AVP transport (parameter SDP m=audio) according to RFC 1890 [14] and RFC 2327 [11].				

Table 19: Mapping of SIP to TCC_OrigCallSetup response from CFE1 to CallingUser

TCC_OrigCallSetup *response/confirmation*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Call Identifier	M	Alphanumeric handle	Call-ID header	Complete call leg (Call-ID, To, From) should better identified the call
Codec	O (notes 1 and 2)	List of possible codecs	SDP parameter encoding name in a=rtpmap: lines	only one a=rtmap SDP line shall be present with a port set to a non null value in the m = SDP line.
Transcode count	O (note 3)	Number of codec transcoding	Number of SDP lines starting with a=rtpmap: and with different encoding name set	The port shall be set to zero in the m = SDP line.
Next Domain Egress point	O	Network specific address	SDP parameter c= and port set in SDP parameter m=audio	A SDP parameter a = has to be set to sendrcv value.
Result	M	- Call established - Rejection cause - Transport not available - Called user busy - Requested QoS not available - Called user unknown - No compatible codec available - Invalid ticket	-200 OK -603 Decline -486 Busy here -488 Not acceptable Here -404 Not found -415 Unsupported Media Type -401 Unauthorized	
NOTE 1: The list of codecs shall be limited to a single entry in the response. NOTE 2: This element shall be included if the result of the request is 'Call established'. NOTE 3: This element shall be included if the result of the request is 'No compatible codec available'.				

Table 20: Mapping of SIP to TCC_CallRelease request from CallingUser to CFE1

TCC_CallRelease *request/indication*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Call Identifier	M	Alphanumeric handle	Call-ID header	Complete call leg (Call-ID, To, From) should better identified the call
CauseCode	M	- UserInitiated - network Initiated	None	In BYE request always set to UserInitiated In CANCEL request it depends of the context

Table 21: Mapping of SIP to TCC_CallRelease response from CFE1 to CallingUser

TCC_CallRelease *response/confirmation*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Call Identifier	M	Alphanumeric handle	Call-ID header	Complete call leg (Call-ID, To, From) should better identified the call
Result	M	- Successful - Failed	-200 OK -4XX to 6XX	

Table 22: Mapping of SIP to TCC_CallAlerting request from CallingUser to CFE1

TCC_CallAlerting *request/indication*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Call Identifier	M	Alphanumeric handle	Call-ID header	Complete call leg (Call-ID, To, From) should better identified the call

7.4.2 Relationship rf, ri (CFE3/CFE6/CFE9)

Table 23: Mapping of SIP to NwCallSetup request exchanged between CFE3/CFE6/CFE9

NwCallSetup *request/indication*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Call Identifier	M	Alphanumeric handle	Call-ID header	Complete call leg (Call-ID, To, From) should better identified the call
Calling user ID restriction	M	Available/unavailable	None	Always set to available in SIP
Calling user ID	O (note 1)	TIPHON user name	SIP-URI contains in From header	Always present
Called user ID	M	TIPHON user name	SIP-URI contains in To header	
PreviousDomainEgresspoint	M	Network specific address	SDP parameter c= and port set in SDP parameter m=audio	
BearerIdentifier	M	Alphanumeric 'handle'	SDP session id set in o=	S=<session name> SDP parameter is not used in SIP
Transport QoSparameter	M	TransportParams	SDP parameters	a=quality:<quality> SDP line could be used
maximumDelay	M	MicroSeconds	None	
maxDelayVariation	M	MicroSeconds	None	
maxMeanPacketLoss	M	PercentX1000	None	
Transportparametersqualifier	M	Enumerated: totalRemainingBudget, budgetAvailableForDomain	None	
TrafficDescriptor	M	TrafficDesc	SDP parameters	Derived to a=ptime:
peakFrameRate	M	Integer	None	
maxFrameLength	M	Integer	None	
Codec	M	List of possible codecs	SDP parameter encoding name in a=rtpmap: lines	note 4
Transcode count	M	Number of codec transcoding	Number of SDP lines starting with a=rtpmap: and with different encoding name set	note 4
Destination Service domain	O (note 2)	Domain address	It can be a Route header field or domain set in TO header field	It is not necessary an IP address
Calling User Access Point	O (note 3)	Network specific address	In Via header field	Always present in SIP
Routing number	O (note 2)	Domain address	Domain set in Request-URI	It is not necessary an IP address

NOTE 1: Shall be present if 'Calling User ID restriction' information element is set to value 'available'.
NOTE 2: This element is available only if by some means routing information or destination network domain can be determined. If so, this information may simplify route calculations in other functional groups.
NOTE 3: The 'Calling User Access Point' may be provided to support the routing decision.
NOTE 4: For a RTP/AVP transport (parameter SDP m=) according to RFC 1890 [14] and RFC 2327 [11].

Table 24: Mapping of SIP to NwCallSetup response exchanged between CFE3/CFE6/CFE9

NwCallSetup *response/confirmation*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Call Identifier	M	Alphanumeric handle	Call-ID header	Complete call leg (Call-ID, To, From) should better identified the call
Next Domain Egress point	M	Network specific address	SDP parameter c= and port set in SDP parameter m=audio	A SDP parameter a= has to be set to sendrcv value.
Codec	O (notes 1 and 2)	List of possible codecs	SDP parameter encoding name in a=rtpmap: lines	only one a=rtmap SDP line shall be present with a port set to a non null value in the m= SDP line.
Transcode count	O (note 3)	Number of codec transcoding	Number of SDP lines starting with a=rtpmap: and with different encoding name set	The port shall be set to zero in the m= SDP line.
Result	M	- Call established - Rejection cause - Insufficient resources - Called user busy - Transport not available - Requested QoS not available - Called user unknown - No compatible codec available	-200 OK -603 Decline -486 Busy here -488 Not acceptable Here -404 Not found -415 Unsupported Media Type	
NOTE 1: The list of codecs shall be limited to a single entry in the response. NOTE 2: This element shall be included if the result of the request is 'Call established'. NOTE 3: This element shall be included if the result of the request is 'No compatible codec available'. NOTE 4: For a RTP/AVP transport (parameter SDP m=) according to RFC 1890 [14] and RFC 2327 [11].				

Table 25: Mapping of SIP to NwCallRelease request exchanged between CFE3/CFE6/CFE9

NwCallRelease *request/indication*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Call Identifier	M	Alphanumeric handle	Call-ID header	Complete call leg (Call-ID, To, From) should better identified the call
CauseCode	M	- UserInitiated - network Initiated	None	In BYE request always set to UserInitiated In CANCEL request it depends of the context

Table 26: Mapping of SIP to NwCallRelease response exchanged between CFE3/CFE6/CFE9

NwCallRelease *response/confirmation*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Call Identifier	M	Alphanumeric handle	Call-ID header	Complete call leg (Call-ID, To, From) should better identified the call
Result	M	- Successful - Failed	-200 OK -4XX to 6XX	

Table 27: Mapping of SIP to NwCallAlerting request exchanged between CFE3/CFE6/CFE9

NwCallAlerting *request/indication*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Call Identifier	M	Alphanumeric handle	Call-ID header	Complete call leg (Call-ID, To, From) should better identified the call

7.4.3 Relationship rl (CFE11/CalledUser)

Table 28: Mapping of SIP to TCC_DestCallSetup request exchanged from CFE11 to CalledUser

TCC_DestCallSetup *request/indication*					
TIPHON				**SIP**	
Information element		**Status**	**Value**	**Mapping**	**Notes**
Call Identifier		M	Alphanumeric handle	Call-ID header	Complete call leg (Call-ID, To, From) should better identified the call
Called user ID		M	TIPHON user name	SIP-URI contains in To header	
Calling user ID		O (note 1)	TIPHON user name	SIP-URI contains in From header	Always present
Transport QoSparameter		M	TransportParams	SDP parameters	a=quality:<quality> SDP line could be used
	maximumDelay	M	MicroSeconds	None	
	maxDelayVariation	M	MicroSeconds	None	
	maxMeanPacketLoss	M	PercentX1000	None	
Codec		M	List of possible codecs	SDP parameter encoding name in a=rtpmap: lines	note 2
Transcode count		M	Number of codec transcoding	Number of SDP lines starting with a=rtpmap: and with different encoding name set	note 2
Previous Domain Egress point		M	Network specific address	SDP parameter c= and port set in SDP parameter m=audio	A SDP parameter a= has to be set to sendrcv value. Because only one port can be set, one m=audio line shall be contained in the SDP offer.
NOTE 1: Shall be present if 'Calling User ID restriction' information element is set to value 'available'. NOTE 2: For a RTP/AVP transport (parameter SDP m=) according to RFC 1890 [14] and RFC 2327 [11].					

Table 29: Mapping of SIP to TCC_DestCallSetup response exchanged from CalledUser to CFE11

TCC_DestCallSetup *response/confirmation*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Call Identifier	M	Alphanumeric handle	Call-ID header	Complete call leg (Call-ID, To, From) should better identified the call
Codec	O (notes 1 and 2)	List of possible codecs	SDP parameter encoding name in a=rtpmap: lines	note 3
Transcode count	O (note 4)	Number of codec transcoding	Number of SDP lines starting with a=rtpmap: and with different encoding name set	note 3
Next Domain Egress point	O	Network specific address	SDP parameter c= and port set in SDP parameter m=audio	A SDP parameter a= has to be set to sendrcv value.
Result	M	- Call established - Rejection cause - Called user busy - No compatible codec available	-200 OK -486 Busy here -415 Unsupported Media Type	
NOTE 1: The list of codecs shall be limited to a single entry in the response. NOTE 2: This element shall be included if the result of the request is 'Call established'. NOTE 3: This element shall be included if the result of the request is 'Call established'. NOTE 4: For a RTP/AVP transport (parameter SDP m=) according to RFC 1890 [14] and RFC 2327 [11].				

Table 30: Mapping of SIP to TCC_CallRelease request exchanged between CFE11/CalledUser

TCC_CallRelease *request/indication*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Call Identifier	M	Alphanumeric handle	Call-ID header	Complete call leg (Call-ID, To, From) should better identified the call
CauseCode	M	- UserInitiated - network Initiated	None	In BYE request always set to UserInitiated In CANCEL request it depends of the context

Table 31: Mapping of SIP to TCC_CallRelease response exchanged between CalledUser/CFE11

TCC_CallRelease *response/confirmation*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Call Identifier	M	Alphanumeric handle	Call-ID header	Complete call leg (Call-ID, To, From) should better identified the call
Result	M	- Successful - Failed	-200 OK -4XX to 6XX	

7.5 Simple call functional entity actions mapping

Actions at CFE5, CFE8 and CFE10 concerning media reservation are out of scope. Proxies in charge of calling and called User have been considered to be stateful.

Table 32: Mapping of SIP to Simple call action at CFE1, CFE2

TIPHON Action number	SIP behaviour
101, 201, 203	Check the proxy authorization header
102	Transmits to the client part the received INVITE
103	Forwards provisional response
104	Forwards 2XX response
105	Forwards the BYE/CANCEL
106	None
107	Forwards the 404 response
108	Sends a 401 response
109	Forwards the 603 response
110	Forwards the 488 response
111	Forwards the 415 response
112	Forwards the 486 response
113, 114	Transmits to the server part the CANCEL, answers to the UA to its CANCEL request with 200 Ok and to its INVITE with a 487 (Request terminated)

Table 33: Mapping of SIP to Simple call action at CFE3, CFE4

TIPHON Action number	SIP behaviour
301,401	Build request URI, update route set according to the received INVITE and eventually DNS information.
302	Out of scope (RSVP), proxy will have to record itself in a record-route header.
303	Forwards the INVITE.
304	Transmits to the server part the provisional response.
305	Out of scope (RSVP) the media connect in SIP will be done one receipt of the ACK.
306	Transmits to the server part 2XX response.
307	The received BYE is transmitted to a client part. The release of the resource is out of scope(RSVP).
308	Forwards the BYE response.
309	Transmits a 404 answer to server part.
310,311	Transmits the 603 response to server part. The release of the resource is out of scope(RSVP).
312	Transmits the 415 response to server part. The release of the resource is out of scope(RSVP).
313	Transmits the 486 response to server part. The release of the resource is out of scope(RSVP).
314	Forwards the CANCEL request. The release of the resource is out of scope (RSVP).

Table 34: Mapping of SIP to Simple call action at CFE6, CFE7

TIPHON Action number	SIP behaviour
601, 701	Build request URI, update route set according to the received INVITE and eventually DNS information.
602	Out of scope (RSVP), proxy will have record itself in a record-route.
603	Sends the updated INVITE to the proxy of the destination user.
604	Forwards provisional response.
605	Out of scope (RSVP), proxy will have record itself in a record-route. The media connect in SIP will be done one receipt of the ACK.
606	Forwards 2XX response.
607	Forwards the BYE. The release of the resource is out of scope(RSVP).
608	Forwards the BYE/CANCEL response.
609	Sends/Forwards a 603 response to CFE3.
610	Sends/Forwards a 488 response to CFE3.
611	Sends/Forwards a 415 response to CFE3.
612	Sends/Forwards a 404 response to CFE3.
613	Forwards the CANCEL The release of the resource is out of scope (RSVP).
614	Forwards the CANCEL response.
615	Update SDP body, sends an ACK to CFE9 an sends a re-INVITE.
616	Sends a 2XX answer.

Table 35: Mapping of SIP to Simple call action at CFE9

TIPHON Action number	SIP behaviour
901	Out of scope (RSVP).
902	Transmits to the client part the received INVITE. An Alerting could not be sent until a provisional response will be received from the User Agent.
903	Out of scope (RSVP).
904	Forwards 2XX response.
905	Forwards the BYE. The release of the resource is out of scope(RSVP).
906	None.
907	Sends/Forwards a 603 response to CFE6.
908	Sends/Forwards a 415 response to CFE6.
909	Sends/Forwards a 486 response to CFE6.
910	Transmits to the client part the received CANCEL that will wait the answer before an answer will be sent to CFE6. The release of the resource is out of scope (RSVP).
911	None.

Table 36: Mapping of SIP to Simple call action at CFE11

TIPHON Action number	SIP behaviour
1101	Forwards the INVITE to the UA
1102	Transmits to the server part the INVITE 2XX response
1103	The received BYE is transmitted to a client part.
1104	The proxy will have to wait the final answer o the BYE before sending any confirmation.
1105	None
1106	Transmits a 415/486 answer to server part.
1107	Forwards the CANCEL request
1108	Transmits to the server part the CANCEL 2XX response and wait for a final response from the UA concerning the INVITE to be acknowledged.

7.6 Timers

The timer defined in meta-protocol for simple call concerns resource reservation which is out of scope in the present document.

7.7 Conclusion

Resource reservation is not in SIP scope, other protocols like RSVP will have to be studied.

The matching of information element for QoS and traffic descriptors to SIP is unsuccessful. However, those parameters can be partially expressed depending of the RTP/AVP profile chosen in SDP.

Alerting is not supported as it is described in the Meta-Protocol, the closer behaviour in SIP is answering with provisional responses but they are sent by the UA and not by the Proxy that forwards them only.

In case of a successful call set-up, call release cannot be confirmed before the remote UA answer. Additional release flows action between CFE9 and CFE11 (action 911) or CFE1 and CFE3 (action 106) cannot be mapped. When the call clear occurs before the call-setup is complete, the UA will receive a final 487 answer to its INVITE request that has no equivalence in the meta protocol flow. The stateful proxy will not wait for the answer from the other UA to send both final responses to the CANCEL and to the INVITE.

The media path is established on receipt of an ACK that has no equivalence in the meta-protocol flow. Moreover, all final responses (even unsuccessful) to an INVITE have to be confirmed with an ACK.

A request in SIP can be forked. Several forwarding destinations can be chosen, and be answered with several response. This is not reflected in the meta-protocol. A one to one mapping at flow level cannot be done.

8 Media Control service

8.1 Introduction

The goal of this clause is to try to map as far as possible the media control service defined in the Meta-protocol document TS 101 882-4 (see bibliography) with SDP (RFC 2327 [11]). RSVP or other equivalent protocol have been considered out of the scope of this clause.

As defined in TS 101 882-4 (see bibliography), the Media Control (MC) service establishes the media elements required to support both call and bearer. It is used to establish a QoS controlled transport capability in accordance with the QoS class identified by the call control meta-protocol.

MC does the following:

- maintains the media state;
- establishes and releases media elements.

SDP allows to describe and negotiate parameters concerning the media. SDP session descriptions are included the body part of SIP messages while the session is established. SDP protocol does not include signalling protocol and only a static mapping for data descriptions has been done.

8.2 Media Control functional entities mapping

Table 37: Mapping of SIP entities to TIPHON Media Control functional entities

TIPHON functional element		SIP entities
Identity	**TIPHON Description**	
Call Control Agent	Request reservation, allocation or release of specific media stream capabilities	Part of UA, Proxy
MFE1	Media control coordination	Part of UA, Proxy

8.3 Media Control information flow Mapping

The mapping is done with the SDP description contained in the body part of SIP a message. Because there is no message information described while the reservation is done, only the request could be statistically described and derived from the SIP message.

8.3.1 Relationship ra (CCA/MFE1)

Because there is no exchange described while the reservation is done, only the MediaReservation request could be statistically described and is concerned by SDP parameters set in the SIP message.

Table 38: Mapping of SIP to MediaReservation request from CCA to MFE1

MediaReservation *request/indication*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Session Handle	M	Alphanumeric 'handle'	SDP session id set in o=	
Media	M	enumerated	SDP parameter m=audio	
Media resource	M			
mediaResourceHandle	O	Integer	SDP payload type set in m= and reused in a=	
rxFlowDescriptor	O			Attribute set when a=recvonly
FlowDescriptorHandle	O	Integer	payload type set in a=	
codecDescriptor	M		derived from RTP/AVP profile set in a=rtmap:xxx	
codecID	M	VisibleString	None	
framesPerPacket	M	Integer	None	
SilenceSupressionEnabled	M	BOOLEAN	None	
codecSpecificParameters	M	VisibleString	None	
transportDescriptors	M			
TransportQosParams	M		could be derived from SDP parameter a= quality:<quality>	
maximumDelay	M		None	
maxDelayVariation	M		None	
maxMeanPacketLoss	M		None	
TraficDescr	M		derived from SDP parameter a=ptime	
peakFrameRate	M	Integer	None	
framesPerPacket	M		None	
ingressAddress	M	E164/E212/IP address	Address set in SDP Parameter c= plus port set in m=audio of the initial offer	
destTransportDomain	M	E164/E212/IP address	Address set in SDP Parameter c= plus port set in m=audio of the final answer	
txFlowDescriptor	O			Attribute set when a=sendonly
FlowDescriptorHandle	O	Integer	payload type set in a=	
codecDescriptor	M		derived from RTP/AVP profile set in a=rtmap:xxx	
codecID	M	VisibleString	None	
framesPerPacket	M	Integer	None	
SilenceSupressionEnabled	M	BOOLEAN	None	
codecSpecificParameters	M	VisibleString	None	
transportDescriptors	M			
TransportQosParams	M			
maximumDelay	M		None	

MediaReservation *request/indication*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
maxDelayVariation	M		None	
maxMeanPacketLoss	M		None	
TraficDescr	M		derived from SDP parameter a=ptime	
peakFrameRate	M	Integer	None	
framesPerPacket	M		None	
ingressAddress	M	E164/E212/IP address	Address set in SDP parameter c= plus port set in m=audio of the initial offer	
destTransportDomain	M	E164/E212/IP address	Address set in SDP parameter c= plus port set in m=audio of the final answer	
connectionPriority	M	Normal, Emergency	None	

8.4 Conclusion

Very few elements from the media Control meta-protocol can be mapped. Other protocols like RSVP will need further study.

9 Transport

9.1 Introduction

The Transport meta-protocol TS 101 882-5 [7] is not stable yet and this clause will need further study.

10 Supplementary services

The meta-protocol for supplementary services has not been defined yet and the mapping with SIP could not be done.

No supplementary service on its own is defined in SIP, other SIP extension IETF RFC will have to be considered.

11 Control of end-to-end Quality of Service

11.1 Introduction

As stated in TS 102 024-3 [3], end-to-end QoS Signalling is used within a TIPHON network to ensure that a caller is provided with an end-to-end connection having at least the QoS class subscribed to or a lower QoS class if this is acceptable to the user. A QoS level may either be requested explicitly by the user on a call-by-call basis or may be predefined as part of the user's subscription. Additionally, the caller may be able to take specific actions if the QoS moves outside the accepted level during an established call.

For each element in each of the QoS Signalling information flows, the tables identify where and how the information can be obtained or sent in the Session Initiation Protocol (SIP) (RFC 3261 [9]) and its associated protocols, the Session Description Protocol (SDP) (RFC 2327 [11]), and the Real-Time Protocol Audio-Video Profile (RTP/AVP) (RFC 1890 [14]). The underlying architectural model of SIP is simpler than the TIPHON model as there is no provision for guaranteed QoS.

11.2 Control of end-to-end Quality of Service functional entities mapping

Table 39: Mapping of SIP entities to TIPHON control of end-to-end Quality of Service entities

TIPHON functional element		SIP entities	
Identity	**TIPHON Description**	**User at Home**	**Roaming User**
Calling User	The application at the calling user"s terminal which instigates the service request.	UA Client	UA Client
QFE1	The service agent that processes the calling user"s request for end-to-end QoS signalling	Application in the home stateful Proxy in the same current domain of the UA (Server Part)	Application in the home stateful Proxy (Server Part)
QFE2	The originating QoS coordination function. This QFE is responsible for negotiating and establishing a particular QoS on behalf of the calling user.	Application in the home stateful Proxy in the same current domain of the UA (Client Part)	Application in the home stateful Proxy (Client Part)
QFE3	The terminating QoS coordination function. This QFE is responsible for establishing a particular QoS on behalf of the called user.	Application in the stateful Proxy of called Domain	Application in the stateful Proxy of called Domain
QFE4	The service agent that processes an incoming call to the called user	stateful Proxy of called Domain	stateful Proxy of called Domain
QFE5	The QoS policy control function associated with the calling user"s service provider	Application in the home stateful Proxy in the same current domain of the UA (Registrar part)	Application in the home stateful Proxy (Registrar part)
QFE6	The originating call routing function, providing routing information and number/address translations;	DNS server/Location Server	SIP routing rules, DNS server, Location Server
QFE7	The transport coordination function serving the called user	stateful Proxy of called Domain	stateful Proxy of called Domain
QFE8	An intervening QoS coordination function. This QFE is responsible for establishing a particular QoS within an intervening domain.	Application in an intermediate stateful Proxy	Application in an intermediate stateful Proxy
QFE9	An intervening transport coordination function.	Intermediate stateful Proxy	Intermediate stateful Proxy
Called User	The application in the called user"s terminal at which the service request is terminated	UA Server	UA Server

11.3 Control of end-to-end Quality of Service flows mapping

SIP and its associated standards are intended for providing communications without a guarantee of QoS. As a consequence, the underlying model is different from the TIPHON model. SIP assumes a direct, but uncontrolled media path to the destination whereas TIPHON assumes linked transport domains carefully controlled by service domains to ensure that sufficient resources are available that the desired QoS can be achieved. There is, therefore, no functional equivalence in SIP or SDP to the messages that pass between a TIPHON service domain and the corresponding transport domain (TRMReserve, TRMConnect and TRMRelease) and, thus, no mapping of meta-protocol information elements to SIP, SDP or RTP/AVP signals is possible.

Table 40: Mapping of SIP messages to TIPHON Control of end-to-end Quality of service MPMUs

TIPHON message		RelationShip ID	SIP messages
OrigQoSEstab	request	Calling User->QFE1	SDP included in INVITE request
	response	QFE1<- Calling User	SDP included in Final responses
QoSEstab	request	QFE1->QFE2 QFE3->QFE4	None
	response	QFE2<-QFE1 QFE4<-QFE3	None
QoSEstab	request	QFE2->QFE8 QFE8->QFE3	SDP included in INVITE request
	response	QFE8<-QFE2 QFE3<-QFE8	SDP included in Final responses
DestQoSEstab	request	QFE4-> Called User	SDP included in INVITE request
	response	Called User<-QFE4	SDP included in Final responses
QoSPolicy		QFE1<->QFE5	None/the Common Open Policy Service (COPS) protocol [17]
TRMReserve		QFE1<->QFE6 QFE8<->QFE9 QFE3<->QFE7	None
TRMConnect		QFE1<->QFE6 QFE8<->QFE9 QFE3<->QFE7	None
TRMRelease		QFE1<->QFE6 QFE8<->QFE9 QFE3<->QFE7	None

11.4 Control of end-to-end Quality of Service information flow data mapping

11.4.1 Relationship ra (CallingUser/QFE1)

Table 41: Mapping of SIP to OrigQoSEstab request from CallingUser to QFE1

OrigQoSEstab *request/indication*				
TIPHON			**SIP/SDP/RTP/AVP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
QoSServiceClass	M	enumerated Value	The suggested attribute for quality *(a=quality:<quality>)* in SDP offers an integer range of 0 to 10. These can be mapped thus: 0 TIPHON Class 1 1 TIPHON Class 2A 2 TIPHON Class 2M 3 TIPHON Class 2H 4 TIPHON Class 3 5 Predefined 6 - 10 Non-standardized QoS classes	The 'quality' attribute is intended primarily for video media streams but there is nothing in RFC 2327 [11] which would prevent it being used for voice QoS Although the range suggested for the SDP quality attribute is 0 to 10, there is no reason why this could not be extended to the TIPHON range of 0 to 255
Called user ID	M	TIPHON user name	SIP-URI contains in To header	
Codec	M	- List of possible codecs - Codec type - Frames per packet	SDP parameter encoding name in a=rtpmap: lines or SDP *media announcements* (m) sub-field, *media formats*. This can carry a list of RTP/AVP codes for available codec types. For example, to use G.711, it is necessary to select the µ-Law PCM code '0', as follows: *m=audio 49232 RTP/AVP 0* No equivalent to Frames per packet	As a default, the Frames per packet should be set to the value 20 in this case (see note)
NOTE: The value of 20 G.711 samples per packet is not entirely arbitrary but is based on the common use of 20 or 30 in existing devices which packetize G.711 sample streams. However, there appears to be no published research on determining the optimum value.				

Table 42: Mapping of SIP to OrigQoSEstab response from QFE1 to CallingUser

OrigQoSEstab *response/confirmation*				
TIPHON			**SIP/SDP/RTP/AVP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Codec	O (notes 1 and 2)	List of possible codecs	SDP parameter encoding name in a=rtpmap: lines	only one a=rtmap SDP line shall be present with a port set to a non null value in the m= SDP line.
Result	M	- End-to-end QoS Established - with requested QoS - Rejection cause - Requested QoS not available - Called user unknown - No compatible codec available - Policy Rejection	SIP INVITE response 200 SIP INVITE request failure - 406: Not acceptable - 404: Not found - 415: Unsupported media type - 401 not authorized	
NOTE 1: The list of codecs shall be limited to a single entry in the response. NOTE 2: This element shall be included if the Result element is set to 'end-to-end QoS Established'.				

11.4.2 Relationship rc, rd (QFE2/QFE8/QFE3)

Table 43: Mapping of SIP to QoSEstab request exchanged between QFE2/QFE8/QFE3

QoSEstab *request/indication*					
TIPHON				**SIP**	
Information element		**Status**	**Value**	**Mapping**	**Notes**
Calling user ID		M	TIPHON user name	SIP-URI contains in From header	For simple mapping to TIPHON user name, the From field should be formulated as a telephone-url as specified in RFC 2806 [16]
Called user ID		M	TIPHON user name	SIP-URI contains in To header	For simple mapping to TIPHON user name, the To field should be formulated as a telephone-url as specified in RFC 2806 [16]
Transport QoSparameter		M	TransportParams	SDP parameters	a=quality:<quality> SDP line could be used
	maximumDelay	M	MicroSeconds	None	
	maxDelayVariation	M	MicroSeconds	None	
	maxMeanPacketLoss	M	PercentX1000	None	
Transportparametersqualifier		M	Enumerated: totalRemainingBudget, budgetAvailableForDomain	None	This information could be carried in a TIPHON-defined *attribute* (a) sub-field in SDP

QoSEstab *request/indication*					
TIPHON				**SIP**	
Information element		**Status**	**Value**	**Mapping**	**Notes**
TrafficDescriptor		M	TrafficDesc	SDP parameters	Derived to a=ptime:
	peakFrameRate	M	Integer	None	
	maxFrameLength	M	Integer	None	
Codec		M	List of possible codecs - Codec type - Frames per packet	SDP parameter encoding name in a=rtpmap: lines	SDP *media announcements* (m) sub-field, *media formats*. This can carry a list of RTP/AVP codes for available codec types. For example, to use G.711, it is necessary to select the µ-Law PCM code '0', as follows: *m=audio 49232 RTP/AVP 0* The Frames per packet information could be carried as a TIPHON-defined *attribute* (a) sub-field in SDP (see note)
Destination Service domain		O	Domain address	Connection address sub-field in the SDP connection data © field	Current edition of SDP supports only IPV4 addresses
NOTE: For a RTP/AVP transport (parameter SDP m=) according to RFC 1890 [14] and RFC 2327 [11].					

Table 44: Mapping of SIP to QoSEstab response exchanged between QFE2/QFE8/QFE3

QoSEstab *response/confirmation*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Codec	O (notes 1 and 2)	List of possible codecs - Codec type - Frames per packet	SDP parameter encoding name in a=rtpmap: lines	only one a=rtmap SDP line shall be present with a port set to a non null value in the m= SDP line.
Result	M	- Requested QoS available - Rejection cause - Requested QoS not available - Called user unknown - No compatible codec available	- 200 OK - 603 Decline - 404 not found - 415: Unsupported media type	
NOTE 1: The list of codecs shall be limited to a single entry in the response. NOTE 2: This element shall be included if the result of the request is 'Call established'.				

11.4.3 Relationship rf (QFE4/CalledUser)

Table 45: Mapping of SIP to DestQoSEstab request exchanged between QFE4/CalledUser

DestQoSEstab *request/indication*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Calling user ID	M	TIPHON user name	SIP-URI contains in From header	For simple mapping to TIPHON user name, the From field should be formulated as a telephone-url as specified in RFC 2806 [16]
Transport QoSparameter	M	TransportParams	SDP parameters	a=quality:<quality> SDP line could be used
maximumDelay	M	MicroSeconds	None	
maxDelayVariation	M	MicroSeconds	None	
maxMeanPacketLoss	M	PercentX1000	None	
Codec	M	List of possible codecs - Codec type - Frames per packet	SDP parameter encoding name in a=rtpmap: lines	SDP *media announcements* (m) sub-field, *media formats*. This can carry a list of RTP/AVP codes for available codec types. For example, to use G.711, it is necessary to select the µ-Law PCM code '0', as follows: *m=audio 49232 RTP/AVP 0* The Frames per packet information could be carried as a TIPHON-defined *attribute* (a) sub-field in SDP (see note)
NOTE: For a RTP/AVP transport (parameter SDP m=) according to RFC 1890 [14] and RFC 2327 [11].				

Table 46: Mapping of SIP to DestQoSEstab response exchanged between QFE4/CalledUser

DestQoSEstab *response/confirmation*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Codec	O (notes 1 and 2)	List of possible codecs - Codec type - Frames per packet	SDP parameter encoding name in a=rtpmap: lines	only one a=rtmap SDP line shall be present with a port set to a non null value in the m= SDP line.
Result	M	- Indicated codec selected - Rejection cause: - Codecs not supported	- 200 OK - 415: Unsupported media type	
NOTE 1: The list of codecs shall be limited to a single entry in the response. NOTE 2: This element shall be included if the result of the request is 'Indicated codec selected'.				

11.4.4 Relationship rg (QFE1/QFE5)

The SIP model does not include a policy entity and so there is no equivalent to the QoSPolicy protocol messages. Consequently, it is not possible to make any mapping between the TIPHON meta-protocol and SIP in this area. However, the Common Open Policy Service (COPS) protocol (RFC 2748 [17]) used by RSVP exists specifically for this purpose. Its underlying architectural model is similar to the TIPHON QoS model in that COPS provides communication between a network node and a policy entity referred to as the Policy Decision Point (PDP).

Table 47: Mapping of SIP to QoSPolicy request exchanged between QFE1/QFE5

QoSPolicy *request/indication*					
TIPHON				**SIP**	
Information element		**Status**	**Value**	**Mapping**	**Notes**
Calling user ID		M	TIPHON user name	COPS REQ C-num=3 *In Interface* C-type=1 (IPv4) or 2 (IPv6)	COPS permits only an IPv4 or IPv6 address. The TIPHON user name would need to be converted from an E.164 number before use
Called user ID		M	TIPHON user name	COPS REQ C-num= 4 *Out Interface* C-type=1 (IPv4) or 2 (IPv6)	COPS permits only an IPv4 or IPv6 address. The TIPHON user name would need to be converted from an E.164 number before use
Transport QoSparameter		M	TransportParams	No equivalent	This information could be carried in the *ClientSI* object (C-num = 9, C-type = 1)
	maximumDelay	M	MicroSeconds	None	
	maxDelayVariation	M	MicroSeconds	None	
	maxMeanPacketLoss	M	PercentX1000	None	
QoSServiceClass		M	enumerated Value	No equivalent	This information could be carried in the *ClientSI* object (C-num = 9, C-type = 2)

Table 48: Mapping of SIP to QoSPolicy response exchanged between QFE1/QFE5

QoSPolicy *response/confirmation*				
TIPHON			**SIP**	
Information element	**Status**	**Value**	**Mapping**	**Notes**
Result	M	- Call permitted - Rejection cause - Service not subscribed to - Service currently not available	COPS DEC C-num = 6, C-type = 1; 4 COPS DRQ C-num = 5 C-type = 1; 9 C-type = 1; 7	C-Type 1; 4 = Admit Request C-Type 1, 9 = Unsupported decision C-type 1, 7 = Insufficient resources

11.5 Control of end-to-end Quality of Service functional entity actions mapping

Those actions are application dependant without equivalence in SIP behaviour.

11.6 Timers

The timer defined in the meta-protocol for simple call concerns resource reservation which is out of scope in the present document.

11.7 Conclusion

Although, with some assumptions, it is possible to show how SIP and SDP can be mapped to the TIPHON QoS meta-protocol between users and service domains and between service domains, there is no provision in the current version of the IETF series of standards for any signalling between service domains and transport domains. Since this signalling is fundamental to the provision of guaranteed QoS in the TIPHON model, there is a significant gap in the mappings. To achieve full mapping, there needs to be considerable modifications to the SIP-related protocols. This should include:

1) the clear recognition that there are entities which can at least act as service domains between the calling user and the called user;

2) the addition within the SIP/SDP architecture of transport domains;

3) the addition of a new protocol specification for signalling between service domains and transport domains;

4) the extension of COPS to include fields for carrying QoS Class or Transport QoS Parameters;

5) the extension of SDP to include fields for carrying QoS class, Transport QoS Parameters, the Transport Parameters Qualifier, the Traffic descriptor and Codec lists.

12 Security service

This clause is for further study.

List of figures

List of tables

Annex A (informative): Bibliography

- ETSI TR 101 301: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; Release Definition; TIPHON 4 Definition".
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History

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