SLOVENSKI STANDARD
SIST-TS TS 101 884-1 V4.1.1:2004

01-april-2004

Harmonizacija telekomunikacij in internetnega protokola prek omrežij (TIPHON), 4. izdaja - 1. del: Tehnologija preslikave - Implementacija arhitekture TIPHON z uporabo SIP

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

Ta slovenski standard je istoveten z: **TS 101 884-1 Version 4.1.1**

ICS:

33.020	Telekomunikacije na splošno	Telecommunications in general

SIST-TS TS 101 884-1 V4.1.1:2004 **en**

ETSI TS 101 884-1 V4.1.1 (2003-08)

Technical Specification

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

Reference

RTS/TIPHON-03018-1R4

Keywords

architecture, IP, SIP, telephony, VoIP

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

Important notice

Individual copies of the present document can be downloaded from:
http://www.etsi.org

The present document may be made available in more than one electronic version or in print. In any case of existing or perceived difference in contents between such versions, the reference version is the Portable Document Format (PDF). In case of dispute, the reference shall be the printing on ETSI printers of the PDF version kept on a specific network drive within ETSI Secretariat.

Users of the present document should be aware that the document may be subject to revision or change of status. Information on the current status of this and other ETSI documents is available at
http://portal.etsi.org/tb/status/status.asp

If you find errors in the present document, send your comment to:
editor@etsi.org

Contents

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is publicly available for **ETSI members and non-members**, and can be found in ETSI SR 000 314: *"Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards"*, which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (http://webapp.etsi.org/IPR/home.asp).

Pursuant to the ETSI IPR Policy, no investigation, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in ETSI SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

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.

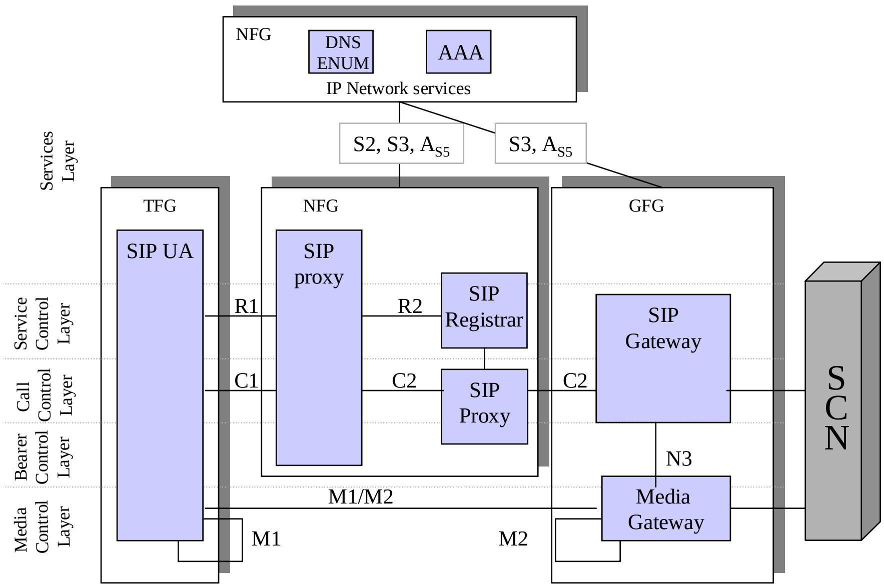
The statefull and stateless proxy maps to the TIPHON service control, call control and bearer layers.

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.