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Harmonizacija telekomunikacij in internetnega protokola prek omrežij (TIPHON) - Študija definicije zahtev - Medsebojno delovanje SIP in H.323

Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); Requirements Definition Study; SIP and H.323 Interworking

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Foreword

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

Introduction

The ETSI Project TIPHON is concerned with the interaction between IP based communication devices and circuit switched networks. The project focuses on voice communication and related multimedia aspects as required for interoperability between IP based networks and other types of networks.

The project has predicated much of its early work on the use of the ITU-T Recommendation H.323 [1] specification since this was the most mature and relevant base specification at the time. The IETF's Internet Multimedia Conferencing Architecture has continued to develop and has started to spawn technologies based upon its signalling and control component - the Session Initiation Protocol (SIP). A SIP Working Group has since been formed within IETF and SIP has been adopted by a number of derivative works, including the IPTEL working group in the IETF. It is therefore appropriate for TIPHON to consider the impact that the introduction of SIP based equipment may have on large-scale public networks.

1 Scope

The present document identifies and defines required service mechanisms to ensure service interoperability for TIPHON which are applicable to release 3. It includes, but is not limited to identifying requirements to interwork between SIP and H.323 administrative domain. The approved delivery from this work item should be used as a common base line for TIPHON and should be used during the whole project life cycle.

2 References

For the purposes of this Technical Report (TR), the following references apply:

[1] ITU-T Recommendation H.323: "Packet-based multimedia communications systems".

[2] RFC 2543 (1999): "SIP: Session Initiation Protocol".

[3] ETSI TS 101 329-3: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); End-to-End Quality of Service in TIPHON Systems; Part 3: Signalling and Control of end-to-end Quality of Service".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

administrative domain: bounded entity within which all encompassed elements are under common ownership, operation and management

endpoint: Entity that can originate and terminate both signalling and media streams. An endpoint can both call and be called. Examples of endpoints include H.323 terminals, SIP User Agents, Gateways, or Multi-party Conference Units.

GateKeeper (GK): H.323 entity on the network which provides address translation and controls access to the network for H.323 terminals, gateways and MCUs. A Gatekeeper may also provide other services such as bandwidth management and gateway location to terminals, gateways and MCUs.

InterWorking Function (IWF): function connecting two networks of differing signalling technology or administrative policies

proxy server: Anetwork element that acts as both a client and server for the purpose of making SIP requests on behalf of other clients. Requests are serviced internally or by passing them on, possibly after translation, to other servers. A proxy interprets and if necessary rewrites a request message before forwarding it.

redirect server: Server that accepts a SIP request, maps the address into zero or more new addresses and returns these addresses to the client. Unlike a proxy server, it does not initiate its own SIP request. Unlike a UAS, it does not accept calls.

registrar: SIP server that accepts REGISTER requests. A registrar is typically co-located with a proxy or re-direct server and MAY offer location services.

Switched Circuit Network (SCN): Telecommunications network, e.g. Public Switched Telephone Network (PSTN), Integrated Services Digital Network (ISDN), and General System for Mobile communications (GSM), that uses circuit-switched technologies for the support of voice calls. The SCN may be a public network or a private network.

telephone call: two-way speech communication between two users by means of terminals connected via network infrastructure

terminal: endpoint other than a gateway or a multipoint control unit

User Agent (UA): application which contains both a UAC and UAS

User Agent Client (UAC): client application that initiates the SIP request

User Agent Server (UAS): Server application that contacts the user when a SIP request is received and that returns a response on behalf of the user. The response accepts, rejects or redirects the request.

3.2 Abbreviations

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

CMS	Call Management Server
CMTS	Call Modem Termination System
CSCF	Call Serve Control Function
DNS	Domain Name Server
GK	H.323 GateKeeper
HFC	Hybrid Fiber Coax
IP	Internet Protocol
ISUP	SS7 ISDN User Part
IWF	InterWorking Function
MT	Mobile Termination
MTA	Multimedia Terminal Adapter
SCN	Switched Circuit Networks
SIP	Session Initiation Protocol
UA	SIP User Agent
UAC	SIP User Client
UAS	SIP User Server
VoIP	Voice over IP

4 Operating modes

H.323 and SIP have a number of declared modes of operation that relate the various functions they identify and define how calls are routed and controlled within their environment. The following clauses identify each mode of operation that is relevant to a potential network context requiring interworking from one protocol technology to another.

4.1 Native H.323 operating modes

H.323 is an architecture for implementing multimedia conferencing over a packet network. It comprises application-layer control protocols that can establish, modify and terminate multimedia sessions or calls. These multimedia sessions may include multimedia conferences, distance learning, Internet telephony and similar applications. It has essentially three possible modes of operation relevant to possible interworking requirements in the context of TIPHON networks.

4.1.1 H.323 peer-to-peer mode

H.323 supports a peer-to-peer mode of operation. In a peer-to-peer architecture, endpoints contact each other directly, without any control or co-ordination from any gatekeeper or intermediate server.

4.1.2 H.323 gatekeeper routed call signalling mode

A gatekeeper may play an active role in mediating call signalling between the calling and called end-points in H.323 networks with gatekeepers. In this environment, a gatekeeper may not only assume responsibility for call routing and authorization on behalf of served endpoints but may also act as the signalling endpoint for calls entering an administrative domain.

4.1.3 H.323 direct call signalling mode

The strict peer-to-peer and gatekeeper routed models may be combined into a hybrid approach referred to as Direct Call Signalling. In this case, gatekeepers provide call routing and authorization while individual endpoints are responsible for establishing and disconnecting calls and media streams directly between each other.

4.1.4 H.323 registration

An H.323 zone is the collection of all terminals, gateways, and Multipoint Control Units managed by a single gatekeeper. Registration is the process by which an endpoint joins an H.323 zone and informs the gatekeeper of its transport and alias addresses. Once established, the registration of an end-point with a specific gatekeeper may need to be refreshed on a periodic basis. An end-point must register with a gatekeeper before it can accept any call attempts.

4.2 Native SIP operating modes

As defined in RFC 2543 [2], the Session Initiation Protocol (SIP) is an application-layer control protocol that can establish, modify and terminate multimedia sessions or calls. These multimedia sessions may include multimedia conferences, distance learning, Internet telephony and similar applications. The most common SIP operation is the invitation. Instead of directly reaching the intended destination, a SIP request may be redirected by Redirect Server or proxied through Proxy Server. Users can also register their location(s) with SIP Registrar.

4.2.1 SIP peer-to-peer

In a peer-to-peer architecture, User Agents (UA) contact each other by sending invitation directly, without any control or co-ordination from any proxy.

4.2.2 SIP proxy routed

SIP messages may be routed via an intermediary known as a proxy server. In such an environment, proxies not only assume responsibility for call routing and authorization on behalf of their endpoints, they may also act as the signalling endpoint for calls into their administrative domain. A proxy server can either be stateful or stateless.

A stateful proxy retains state information concerning both an incoming request and any associated outgoing requests. In contrast, a stateless proxy does not retain any information concerning a received message or its response once an outgoing request has been generated.

4.2.3 SIP with redirect server

SIP redirect servers represent an example of a loosely coupled distributed architecture. In this environment, the redirect server provides the call routing information such that the originating UA first establishes a signalling connection with the redirect server, before being re-directed to the terminating UA.

4.2.4 SIP registration

The SIP REGISTER method allows a client to let a SIP Registrar know at which address or addresses it can be reached. A client may also use it to install call handling features at the server. A SIP Registrar may be collocated with either a proxy or redirect server.

4.3 Recommended modes of operation

4.3.1 H.323 administrative domain

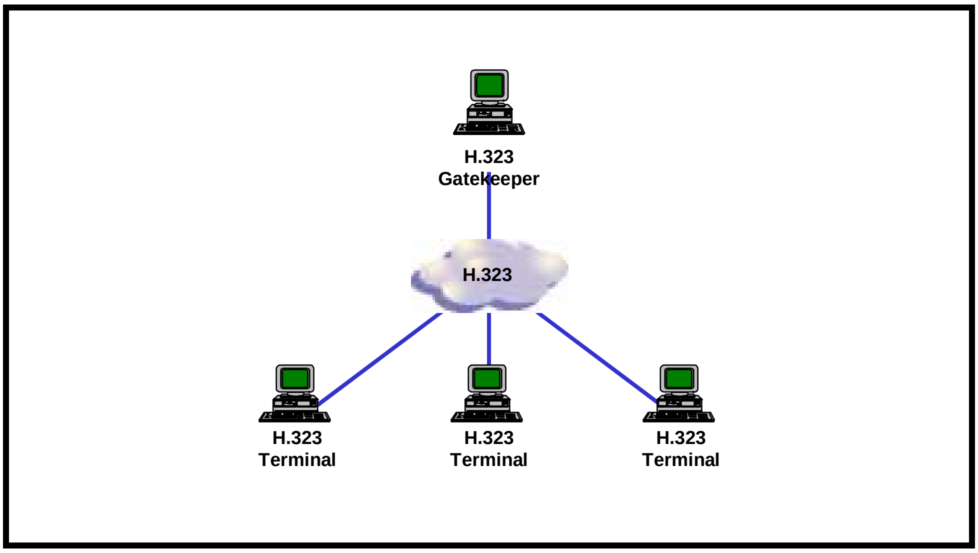

Figure 1: An H.323 administrative domain

While a gatekeeper is an optional element for an H.323 network, it is recommended that TIPHON based H.323 centric networks are deployed with an H.323 Gatekeeper. This should be configured to require registration with all the end points within its administrative domain. It is further recommended that Gatekeeper Routed Call Signalling is used in preference to Direct Call Signalling in order to support enhanced calling features such as availability look-ahead for the called terminal.

4.3.2 SIP administrative domain

As defined in RFC 2543 [2], SIP has no concept of an Administrative Domain. However for practical network engineering and operational reasons consistent with the TIPHON approach to QoS [3], the cocept of a SIP Administrative Domain is introduced. This enables the trust boundaries within which all SIP devices are controlled by a single operator to be delineated. Each SIP Administrative Domain is assumed to contain at least one Registrar. All SIP UAs within that domain must register with the Registrar in order to allow the user's or terminal's address(es) to be advertised within the domain.

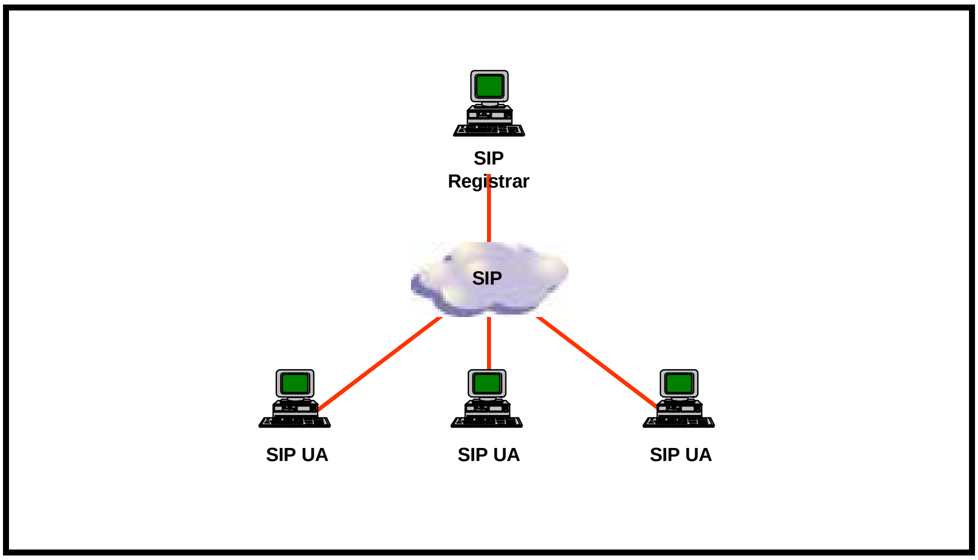

Figure 2: A SIP administrative domain