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Information technology — Telecommunications and information exchange between systems — Corporate Telecommunication Networks — Signalling Interworking between QSIG and SIP — Call Transfer

Technologies de l'information — Télécommunications et échange d'information entre systèmes — Réseaux de télécommunications d'entreprise — Interaction de signalisation entre QSIG et SIP — Transfert d'appel

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Foreword

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International Standards are drafted in accordance with the rules given in the ISO/IEC Directives, Part 2.

The main task of the joint technical committee is to prepare International Standards. Draft International Standards adopted by the joint technical committee are circulated to national bodies for voting. Publication as an International Standard requires approval by at least 75 % of the national bodies casting a vote.

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ISO/IEC 23916 was prepared by Ecma International (as ECMA 361) and was adopted, under a special “fast-track procedure”, by Joint Technical Committee ISO/IEC JTC 1, *Information technology*, in parallel with its approval by national bodies of ISO and IEC.

Introduction

This International Standard is one of a series of Standards defining the interworking of services and signalling protocols deployed in corporate telecommunication networks (CNs) (also known as enterprise networks). The series uses telecommunication concepts as developed by ITU-T and conforms to the framework of International Standards on Open Systems Interconnection as defined by ISO/IEC.

This International Standard specifies call transfer interworking between the Session Initiation Protocol (SIP) and QSIG within corporate telecommunication networks (also known as enterprise networks). SIP is an Internet application-layer control (signalling) protocol for creating, modifying, and terminating sessions with one or more participants. These sessions include, in particular, telephone calls.

This International Standard is based upon the practical experience of member companies and the results of their active and continuous participation in the work of ISO/IEC JTC1, ITU-T, IETF, ETSI and other international and national standardization bodies. It represents a pragmatic and widely based consensus.

Information technology — Telecommunications and information exchange between systems — Corporate Telecommunication Networks — Signalling Interworking between QSIG and SIP — Call Transfer

1 Scope

This International Standard specifies call transfer interworking between the Session Initiation Protocol (SIP) and “QSIG” within corporate telecommunication networks (CN), also known as enterprise networks.

"QSIG" is a signalling protocol that operates between Private Integrated services Network eXchanges (PINX) within a Private Integrated Services Network (PISN). A PISN provides circuit-switched basic services and supplementary services to its users. QSIG is specified in Standards, in particular [1] (call control in support of basic services), [2] (generic functional protocol for the support of supplementary services) and a number of Standards specifying individual supplementary services. Transfer services are specified in [3], [6] and the QSIG signalling protocol in support of these services is specified in [4], [7]. In particular, this signalling protocol signals information about call transfer to the users who are involved.

NOTE The name QSIG was derived from the fact that it is used for signalling at the Q reference point. The Q reference point is a point of demarcation between two PINXs.

SIP is an application layer protocol for establishing, terminating and modifying multimedia sessions. It is typically carried over IP. Telephone calls are considered as a type of multimedia session where just audio is exchanged. SIP is defined in [10].

As the support of telephony within corporate networks evolves from circuit-switched technology to Internet technology, the two technologies will co-exist in many networks for a period, perhaps several years. Therefore there is a need to be able to establish, modify, terminate and transfer sessions involving participants in the SIP network and participants in the QSIG network. Such calls are supported by gateways that perform interworking between SIP and QSIG.

This specification specifies SIP-QSIG signalling interworking for transfer services between a PISN employing QSIG and a corporate IP network employing SIP.

2 Normative references

The following referenced documents are indispensable for the application of this document. For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments) applies.

[1] International Standard ISO/IEC 11572 "Information technology -- Telecommunications and information exchange between systems -- Private Integrated Services Network -- Circuit mode bearer services -- Inter-exchange signalling procedures and protocol" (also published by Ecma as Standard ECMA-143).

[2] International Standard ISO/IEC 11582 "Information technology -- Telecommunications and information exchange between systems -- Private Integrated Services Network -- Generic functional protocol for the support of supplementary services -- Inter-exchange signalling procedures and protocol " (also published by Ecma as Standard ECMA-165).

[3] International Standard ISO/IEC 13865 "Information technology -- Telecommunications and information exchange between systems -- Private Integrated Services Network -- Specification, functional model and information flows -- Call Transfer supplementary service" (also published by Ecma as Standard ECMA-177).

[4] International Standard ISO/IEC 13869 "Information technology -- Telecommunications and information exchange between systems -- Private Integrated Services Network -- Inter-exchange signalling protocol -- Call Transfer supplementary service" (also published by Ecma as Standard ECMA-178).

[5] International Standard ISO/IEC 17343 "Information technology -- Telecommunications and information exchange between systems -- Corporate telecommunication networks -- Signalling interworking between QSIG and SIP -- Basic services" (also published by Ecma as Standard ECMA-339).

[6] International Standard ISO/IEC 19459 "Information technology -- Telecommunications and information exchange between systems -- Private Integrated Services Network -- Specification, functional model and information flows -- Single Step Call Transfer Supplementary Service" (also published by Ecma as Standard ECMA-299).

[7] International Standard ISO/IEC 19460 "Information technology -- Telecommunications and information exchange between systems -- Private Integrated Services Network -- Inter-exchange signalling protocol -- Single Step Call Transfer supplementary service" (also published by Ecma as Standard ECMA-300).

[8] Ecma Technical Report TR/86, "Corporate Telecommunication Networks – User Identification in a SIP/QSIG Environment".

[9] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119.

[10] J. Rosenberg, H. Schulzrinne, et al., "SIP: Session initiation protocol", RFC 3261.

[11] J. Peterson, "A Privacy Mechanism for the Session Initiation Protocol (SIP)", RFC 3323.

[12] R. Sparks, "The Session Initiation Protocol (SIP) REFER Method", RFC 3515.

[13] R. Mahy, B. Biggs, R. Dean, "The Session Initiation Protocol (SIP) "Replaces" Header", RFC 3891.

[14] R. Sparks, "The Session Initiation Protocol (SIP) Referred-By Mechanism", RFC 3892.

[15] R. Sparks, A. Johnston, "Session Initiation Protocol Call Control - Transfer", draft-ietf-sipping-cc-transfer-02 (work in progress).

3 Terminology

In this document, the key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" are to be interpreted as described in RFC 2119 [9] and indicate requirement levels for compliant SIP implementations.

In the interests of keeping the normative text and the diagrams as simple as possible, the QSIG messages in this document implicitly follow QSIG signalling rules of [1] and [2]. For instance, sending a QSIG DISCONNECT message on a call where a QSIG DISCONNECT has already been sent is implicitly forbidden and therefore not mentioned as such in this document.

The figures in this document are provided as examples. They are not normative. In the interests of keeping the diagrams simple, some SIP messages (ACK, PRACK, final responses to BYE and NOTIFY) are not shown.

The following notation is used for call transfer information within QSIG messages:

— xxx.inv - invoke application protocol data unit (APDU) of operation xxx.

— xxx.res - return result APDU of operation xxx.

— xxx.err - return error APDU of operation xxx.

The following abbreviations are used:

— ctActive stands for callTransferActive.

— ctComplete stands for callTransferComplete.

The drawings use the following conventions:

— D1 and D2 are SIP dialogs. CR1 and CR2 are QSIG call references. By convention, D1 is mapped to CR1 and D2 to CR2.

— A SIP message is prefixed by (Dx-y), when it belongs to SIP dialog Dx and is part of SIP transaction y.

— The method or response code of the SIP messages is displayed, as well as the name of SIP header fields that play a role in the interworking functions. Some examples display an "Identity:" information field. It indicates that the local identity information field should be mapped to a real SIP identity information field as described in 7.4.

4 Definitions

For the purposes of this specification, the following definitions apply.

4.1 External definitions

The definitions in [1] and [10] apply as appropriate.

4.2 Other definitions

4.2.1
User A
the served user, i.e. the user requesting Call transfer or Single step call transfer.

4.2.2
User B
a user who is in communication with User A and who will be transferred to User C.

NOTE This definitions differs from [3], in order to use similar conventions for QSIG Call transfer and QSIG Single step call transfer.

4.2.3
User C
the user to whom the call is transferred.

4.2.4
Call transfer
the act of enabling a user (User A) to transform two of that user's calls (at least one of which must be answered) into a new call between the two other users (User B and User C) in the two calls.

NOTE Call transfer is very similar to the "attended transfer" described in [15].

A Call transfer before answer is a Call transfer that occurs before User C answers the call initiated by User A.

4.2.5
Single step call transfer
the act of enabling a served user (User A) to transfer an active call (with User B) to a user (User C) that has no call established either to User A or to User B. On successful completion of Single step call transfer, User B and User C can communicate with each other and User A are no longer involved in a call with User B or User C.

NOTE Single step call transfer is very similar to the "basic transfer" described in [15].

4.2.6
Call transfer by join
the effecting of transfer when User A is a PISN user by joining together the connections of the calls to User B and User C at User A's PINX.

4.2.7
Call transfer by rerouteing
the effecting of transfer by establishing a new connection to replace all or part of the connections of the calls to User B and User C.

4.2.8
Corporate telecommunication Network (CN)
sets of privately-owned or carrier-provided equipment that are located at geographically dispersed locations and are interconnected to provide telecommunication services to a defined group of users.

NOTE 1 A CN can comprise a PISN, a private IP network (intranet) or a combination of the two.

NOTE 2 Also known as enterprise network.

4.2.9
IP network
a network, unless otherwise stated a corporate network, offering connectionless packet-mode services based on the Internet Protocol (IP) as the network layer protocol.

4.2.10
Private Integrated Services Network (PISN)
a CN or part of a CN that employs circuit-switched technology and QSIG signalling.

4.2.11
Private Integrated services Network eXchange (PINX)
a PISN nodal entity comprising switching and call handling functions and supporting QSIG signalling in accordance with [1].

5 Abbreviations and acronyms

APDU	Application Protocol Data Unit
IP	Internet Protocol
PINX	Private Integrated services Network eXchange
PISN	Private Integrated Services Network
SIP	Session Initiation Protocol
UA	User Agent
UAC	User Agent Client

UAS User Agent Server

URI Universal Resource Identifier

6 Background and architecture

The background and architecture of [5] applies. In addition, the interworking function in the protocol model handles interworking for call transfer services. This involves interworking between the QSIG call transfer protocol specified in [4], [7] and SIP [10], including the use of the REFER [12] SIP method and Replaces [13] and Referred-By [14] SIP header fields.

7 Procedures

7.1 Call transfers in QSIG

For the purposes of QSIG, transfers are classified into one of the following types:

— call transfer by join: defined in 4.2.6;

— call transfer by rerouteing: defined in 4.2.7; and

— single step call transfer: defined in 4.2.5.

QSIG Call transfer by rerouteing is out of scope of this document because of its lesser deployment.

QSIG signalling for transfers is based on [2] and comprises the following remote operations:

— ssctInitiate - this confirmed operation is sent by User A's PINX to User B's PINX. It initiates a single step call transfer. It includes a "rerouteingNumber" of User C. It also includes an optional "transferredAddress" of User B, an optional "transferringAddress" of User A, and a optional boolean "awaitConnect" that indicates if the operation return result is expected when the call is connected or when it is alerting User C.

— callTransferActive - this unconfirmed operation is sent to User B. It indicates that answer has taken place following a Call transfer before answer. It includes a "connectedAddress" that identifies the other User that answered the transferred call.

— callTransferComplete - this unconfirmed operation is sent to User B and User C. It indicates that a transfer has been effected. It includes an indication of whether the resulting call is alerting or answered (referred to later in this document as "callStatus"). It includes a "redirectionNumber" that identifies the other User in the transferred call.

— ssctSetup - this confirmed operation is sent by User B to User C. It initiates a new call between User B and User C for the purpose of single step call transfer. It includes an optional "transferringAddress" that indicates who initiated the transfer.

— callTransferUpdate - this optional unconfirmed operation is sent to User B and User C. It provides information known to the network about the remote party.

— subaddressTransfer - this optional unconfirmed operation is sent to User B or to User C. It informs each other of the other party's public ISDN subaddress. It includes a "connectedSubaddress" that identifies the public ISDN subaddress.

7.2 Call transfer in SIP

SIP has the concept of requesting a UA to refer (establish a dialog to) a third party [12]. SIP also has the concept of replacing a dialog by another one [13], and provides a mechanism so that information about the initiator of the REFER request is sent to the third party [14]. The call transfer document gives examples on how to support call transfers using these SIP extensions [15].

7.3 Scope of the interworking functions

7.3.1 QSIG side

The interworking functions provided in the following clauses encompass QSIG Call transfer by join and QSIG Single step call transfer. QSIG Call transfer by rerouteing is out of scope of this document because of its lesser deployment. The functions take into account that User A, B and C can be in the IP network or in the PISN.

7.3.2 SIP side

The interworking functions rely on SIP mechanisms. Thus, the interworking functions of this document make the following assumptions:

— the gateway and the SIP User Agents use globally routable SIP addresses, or use SIP addresses in an environment where they are routable, or will use other future mechanisms that allow global routing,

— it is RECOMMENDED that the gateway and the SIP User Agents use SIP security mechanisms related to authentication and confidentiality.

7.3.3 Discussion over transfer interworking functions

This Clause is not normative and aims at giving explanations concerning the implementation choices of this document.

The QSIG Single step call transfer mechanism is very similar to the "basic transfer" described in [15]. The QSIG Call transfer is also very similar to the "attended transfer" described in [15]. The latter uses the REFER method and the Replaces header. Yet, it is not possible to use this mechanism in all the interworking situations. For instance, if User A in the PISN does not call User B and User C in the SIP network through the same gateway, there is no opportunity to optimize the signalling path by using the REFER method in the IP network. Figure 1 gives an example of such a situation where transfer by join has been performed at user A's PINX. The gateway to user B is unaware that user C is also in the IP network (and even if it were aware, it has no information for building a Replaces header). Similarly the gateway to user C is unaware that user B is in the IP network.

