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

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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 23915 was prepared by Ecma International (as ECMA-360) 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 interworking between the Session Initiation Protocol (SIP) and QSIG within corporate telecommunication networks (also known as enterprise networks) for calls that undergo diversion. 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. QSIG is a signalling protocol for creating, modifying and terminating circuit-switched calls, in particular telephone calls, within Private Integrated Services Networks (PISNs). QSIG is specified in a number of Standards and published also as ISO/IEC International Standards.

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 Diversion

1 Scope

This document specifies signalling interworking between "QSIG" and the Session Initiation Protocol (SIP) in support of call diversion 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. Diversion services are specified in [4] and the QSIG signalling protocol in support of these services is specified in [5]. In particular, this signalling protocol signals information about call diversion to the users involved.

SIP is an application layer protocol for establishing, terminating and modifying multimedia sessions. It is typically carried over IP [8], [10]. Telephone calls are considered as a type of multimedia session where just audio is exchanged. SIP is defined in [11]. An extension to SIP provides history information [14] that can be used to signal information about the retargeting of a request, in particular a call establishment request, as it is routed through a network.

This document specifies signalling interworking for call diversion during the establishment of calls between a PISN employing QSIG and a corporate IP network employing SIP. It covers both the impact on SIP of call diversion in the QSIG network and the impact on QSIG of request retargeting in the SIP network. Signalling interworking for call diversion operates on top of signalling interworking for basic calls, which is specified in [6].

Call diversion interworking between a PISN employing QSIG and a public IP network employing SIP is outside the scope of this specification. However, the functionality specified in this specification is in principle applicable to such a scenario when deployed in conjunction with other relevant functionality (e.g., number translation, security functions, etc.).

This specification is applicable to any interworking unit that can act as a gateway 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 13868 "Information technology -- Telecommunications and information exchange between systems -- Private Integrated Services Network -- Inter-exchange signalling protocol -- Name identification supplementary services" (also published by Ecma as Standard ECMA-164).

[4] International Standard ISO/IEC 13872 "Information technology -- Telecommunications and information exchange between systems -- Private Integrated Services Network -- Specification, functional model and information flows -- Call Diversion supplementary services" (also published by Ecma as Standard ECMA-173).

[5] International Standard ISO/IEC 13873 "Information technology -- Telecommunications and information exchange between systems -- Private Integrated Services Network -- Inter-exchange signalling protocol -- Call Diversion supplementary services" (also published by Ecma as Standard ECMA-174).

[6] 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).

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

[8] J. Postel, "Internet Protocol", RFC 791.

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

[10] S. Deering, R. Hinden, "Internet Protocol, Version 6 (IPv6)", RFC 2460.

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

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

[13] H. Schulzrinne, D. Oran, G. Camarillo, "The Reason Header field for the Session Initiation Protocol (SIP)", RFC 3326.

[14] M. Barnes "An Extension to the Session Initiation Protocol for Request History Information", draft-ietf-sipping-history-info-03 (work in progress).

3 Terms and definitions

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 implementations.

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

3.1 External definitions

The definitions in [1] and [11] apply as appropriate.

3.2 Other definitions

3.2.1
Call diversion
the act of retargeting a call during call establishment by changing the user identity that is used as the basis for routing to the destination.

3.2.2
Call forwarding busy (CFB)
call diversion invoked because the targeted user is busy.

3.2.3
Call forwarding no reply (CFNR)
call diversion invoked because the targeted user fails to reply within a certain time.

3.2.4
Call forwarding unconditional (CFU)
call diversion invoked for reasons other than those leading to CFB or CFNR.

3.2.5
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.

3.2.6
Entity A

the entity that provides information about diversion to user A.

3.2.7
Entity B
the entity that invokes diversion for a call targeted at user B.

3.2.8
Entity C
the entity that provides information about diversion to user C.

3.2.9
Gateway
an entity that performs interworking between a PISN using QSIG and an IP network using SIP.

3.2.10
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.

3.2.11
Leg A
the call segment between entity A and the rerouting entity for a call that undergoes diversion.

3.2.12
Leg B
the call segment between the rerouting entity and entity B for a call that undergoes diversion.

3.2.13
Leg C
the call segment between the rerouting entity and entity C for a call that undergoes diversion.

3.2.14
Private Integrated Services Network (PISN)
a CN or part of a CN that employs circuit-switched technology.

3.2.15
Private Integrated services Network eXchange (PINX)
a PISN nodal entity comprising switching and call handling functions and supporting QSIG signalling in accordance with [1].

3.2.16
Rerouting entity
the entity that performs call rerouting on request from entity B and that provides information about diversion to entity A and entity C.

3.2.17
User A
the calling user of a call that undergoes diversion.

3.2.18
User B
the user on behalf of which call diversion is invoked for an incoming call to that user.

3.2.19
User C
the user to which a call is diverted.

4 Abbreviations and acronyms

APDU	Application Protocol Data Unit
CFB	Call forwarding busy
CFNR	Call forwarding no reply
CFU	Call forwarding unconditional
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

5 Background and architecture for SIP-QSIG interworking

The background and architecture of [6] applies. In addition, the interworking function in the protocol model handles interworking for call diversion services. This involves interworking between the QSIG call diversion protocol specified in [5] and SIP, including the use of SIP request history information as specified in [14].

6 Call diversion

Call diversion, as specified in QSIG and for the purposes of this document, is the act of retargeting a call during call establishment by changing the user identity that is used as the basis for routing to the destination. This can be viewed as being a change of destination user, although in some cases two identities can belong to the same user, e.g., a home number and office number. The three users involved are known as user A (the calling user A), user B (the called user or diverting user) and user C (the diverted-to user).

Reasons for invoking diversion are various and can depend on factors such as the state of the line serving user B, the time of day and the type or identity of user A. It could also be as a result of action by user B in response to the arrival of a call (sometimes known as call deflection). A diversion can occur immediately, i.e. without alerting user B, or after a period of alerting without reply. With the exception of call deflection, diversion requirements must be pre-configured into some equipment acting on behalf of user B, e.g., a telephone, a PINX or a SIP proxy. This could be achieved, for example, by rules-based scripting.

It is often useful or even important that the users involved in a diverted call (user A and user C) are informed of the diversion. This can be particularly important for automata, e.g., for a call diverted to a voice mail system it might be important to indicate to the system that the call has been diverted from user B. However, privacy considerations can sometimes lead to the suppression of this information.

The general model for a call that undergoes diversion is shown in Figure 1. Entity B is the entity that invokes diversion, based on configuration or, in the case of call deflection, on request from user B. The rerouting entity performs call rerouting on instruction from entity B and provides information about the diversion to entity A and entity C. Entity A and entity C handle diversion on behalf of users A and C respectively by providing information about diversion.

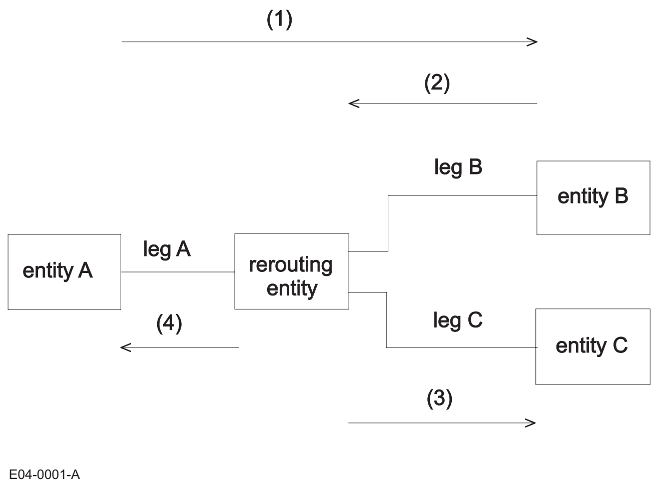

Figure 1 – Logical model for diversion in a QSIG network

From this model it can be seen that there are three call legs:

— leg A between entity A and the rerouting entity (null if these two entities are collocated);

— leg B between entity B and the rerouting entity (null if these two entities are collocated);

— leg C between entity C and the rerouting PINX (null if these two entities are collocated).

Diversion signalling on leg A provides information about diversion to entity A, which can use it to provide information to user A. Diversion signalling on leg B instructs the rerouting entity to carry out rerouting. Diversion signalling on leg C provides information about diversion to entity C, which can use it to provide information to user C.

Figure 1 also illustrates the basic dynamic behaviour:

1. Call establishment from user A as far as entity B.

2. Rerouting request from entity B to the rerouting entity.

3. Rerouted call establishment from the rerouting entity to entity C accompanied by information about the diversion.

4. Information about the diversion from the rerouting entity to entity A.

Diversions can be chained. In this case the rerouted call from the rerouting entity reaches another entity B. The same or a different rerouting entity then reroutes the call towards the new user C.

7 Call diversion in QSIG

Call diversion in QSIG is the act of retargeting a call during call establishment by changing the called party number, which is the user identity used as the basis for routing to the destination. Call diversion in QSIG follows the model described above. Entity A is located in user A's PINX (PINX A), entity B is located in user B's PINX (PINX B) and entity C is located in user C's PINX (PINX C). The rerouting entity is located either at user B's PINX (diversion by forward switching) or at user A's PINX (diversion by rerouting).

Because of potential interactions with other supplementary services, the signalling for which passes transparently through intermediate (Transit) PINXs, the rerouting PINX is constrained to be either PINX B or PINX A. The former case is known as diversion by forward switching, and is analogous to SIP retargeting by a proxy. The latter case is known as diversion by rerouting and is analogous to SIP retargeting by redirection.

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

— call forwarding no reply (CFNR) (forwarding as a result of no user reply after alerting user B for a certain time);

— call forwarding busy (CFB) (forwarding as a result of user B's device being busy); and

— call forwarding unconditional (CFU) (forwarding for reasons other than no reply or busy).

NOTE CFU is not necessarily entirely unconditional, since it can depend on other factors, e.g., time of day.

In common with other supplementary services, QSIG signalling for diversion is based on [2] and comprises the following remote operations:

— callRerouting – this confirmed operation is applicable to leg B and provides a means for PINX B to request the rerouting PINX to reroute a call to user C.

— cfnrDivertedLegFailed – this unconfirmed operation is applicable to leg B and indicates failure to establish call leg C subsequent to accepting a callRerouting operation. cfnrDivertedLegFailed applies only to CFNR (i.e. to diversions after user B has been alerted) and indicates to PINX B that user B should continue to be alerted. For other types of diversion leg B is cleared down as soon as the callRerouting operation is accepted, without waiting to see if the call towards user C can be established.

— divertingLegInformation1 – this unconfirmed operation is applicable to leg A and signals information about the diversion to PINX A, including any privacy requirement of user B to prevent disclosure of diversion information to user A. Note that PINX A can use the information for internal purposes (e.g., call logging) but is trusted not to disclose private information to user A.

— divertingLegInformation2 – this unconfirmed operation is applicable to leg C and signals information about the diversion to PINX C.

— divertingLegInformation3 – this unconfirmed operation is applicable to legs A and C and signals privacy information from PINX C to PINX A. This privacy information provides the possibility for user C to suppress the disclosure of its identity to user A. PINX A must take into account both the privacy information in divertingLegInformation1 and the privacy information in divertingLegInformation3 before disclosing information to user A.

Chained diversions are supported. PINX A receives a divertingLegInformation1 operation for each diversion, but often a divertingLegInformation3 operation only for the final diversion (since this information is not necessarily available until answer). The final PINX C receives a single divertingLegInformation2 operation containing information about the first and last diversions but not intermediate diversions.

8 Call diversion in SIP

Call diversion is not specified for SIP. However, SIP does have the concept of retargeting an INVITE request. This occurs at a proxy, instigated either by the proxy itself or on request from a redirect using a 3xx response. It can also occur at the UAC as a result of a 3xx response from a redirect. Relating this to the model, the rerouting entity for a SIP diversion is the proxy or UAC that retargets the INVITE request. Entity B is either that same proxy or UAC or a redirect that issues a 3xx response. A 3xx response therefore has some synergy with a QSIG callRerouting operation. Entity A is the UAC for the INVITE request and entity C is the UAS of the retargeted-to user.

Retargeting involves changing the Request-URI within the INVITE request, this field being the basis for routing the request.

[11] does not provide signalling support for notifying user A's UA or user C's UA that retargeting has occurred. Additional signalling for this purpose is specified in [14]. This allows a retargeting proxy or UAC to insert a History-Info header into a request when it is forwarded downstream, i.e. on leg C towards entity C. Moreover entity C reflects the received History-Info header back over leg C and leg A towards entity A. In this way, both entity A and entity C receive information about the retarget and can provide this information to their respective users. The History-Info header contains a number of entries, each containing a URI that was a Request-URI at some stage during the routing of the call.

Chained retargets are supported. Entity A and entity C receive information about multiple retargets carried out during the routing of the INVITE request.

History information can be of a sensitive nature, and therefore [14] makes provision for keeping it private. History information subject to privacy must not be passed outside the domain where it originates. Within that domain, the Privacy header [12] with privacy value "history" [14] is used to indicate that either the entire history information or a particular entry is subject to privacy and must not be passed outside the domain.

9 Diversion interworking

9.1 Scenarios for diversion interworking

From the descriptions in clauses 7 and 8 it can be seen that both diversion in QSIG and retargeting, along with the History-Info header, in SIP can be mapped to the call diversion model described in clause 6. Therefore interworking can be described in terms of this model.

Interworking can occur on leg A, on leg B or on leg C. In either case, the rerouting entity can be in either the SIP network or the QSIG network. This leads to 6 interworking scenarios.