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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

ISO (the International Organization for Standardization) and IEC (the International Electrotechnical Commission) form the specialized system for worldwide standardization. National bodies that are members of ISO or IEC participate in the development of International Standards through technical committees established by the respective organization to deal with particular fields of technical activity. ISO and IEC technical committees collaborate in fields of mutual interest. Other international organizations, governmental and non-governmental, in liaison with ISO and IEC, also take part in the work. In the field of information technology, ISO and IEC have established a joint technical committee, ISO/IEC JTC 1.

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.

Attention is drawn to the possibility that some of the elements of this document may be the subject of patent rights. ISO and IEC shall not be held responsible for identifying any or all such patent rights.

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).