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Information technology — Telecommunications and information exchange between systems — Corporate telecommunication networks — Signalling interworking between QSIG and SIP — Basic services

Technologies de l'information — Télécommunications et échange d'information entre systèmes — Réseaux de télécommunications d'entreprise — Signalisation d'interfonctionnement entre QSIG et SIP — Services de base

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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 17343 was prepared by Ecma International (as ECMA-339) 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.

This second edition cancels and replaces the first edition (ISO/IEC 17343:2004), which has been technically revised.

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 defines the signalling protocol interworking for basic services between a Private Integrated Services Network (PISN) and a packet-based private telecommunications network based on the Internet Protocol (IP). It is further assumed that the protocol for the PISN part is QSIG and that the protocol for the IP-based network is SIP. Compared with the first edition of ISO 17343, this second edition includes numerous small changes arising during derived work in the IETF on RFC 4497. This second edition is in full technical alignment with RFC 4497.

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 JTC 1, ITU-T, ETSI and other international and national standardization bodies. It represents a pragmatic and widely based consensus.

In this International Standard, 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 [4] and indicate requirement levels for compliant SIP implementations.

Information technology — Telecommunications and information exchange between systems — Corporate telecommunication networks — Signalling interworking between QSIG and SIP — Basic services

1 Scope

This International Standard specifies signalling interworking between QSIG and the Session Initiation Protocol (SIP) in support of basic services within a corporate telecommunication network (CN) (also known as enterprise network).

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 other Standards; in particular [2] (call control in support of basic services), [3] (generic functional protocol for the support of supplementary services), and a number of standards specifying individual supplementary services.

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 [15], [16]. Telephone calls are considered 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 coexist in many networks for a period, perhaps several years. Therefore, there is a need to be able to establish, modify, and terminate sessions involving a participant in the SIP network and a participant in the QSIG network. Such calls are supported by gateways that perform interworking between SIP and QSIG.

This International Standard specifies SIP-QSIG signalling interworking for basic services that provide a bi-directional transfer capability for speech, DTMF, facsimile, and modem media between a PISN employing QSIG and a corporate IP network employing SIP. Other aspects of interworking, e.g., the use of RTP and SDP, will differ according to the type of media concerned and are outside the scope of this International Standard.

Call-related and call-independent signalling in support of supplementary services is outside the scope of this International Standard, but support for certain supplementary services (e.g., call transfer, call diversion) could be the subject of future work.

Interworking between QSIG and SIP permits a call originating at a user of a PISN to terminate at a user of a corporate IP network, or a call originating at a user of a corporate IP network to terminate at a user of a PISN.

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

This International Standard 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 Conformance

In order to conform to this International Standard, a gateway shall satisfy the requirements identified in the Implementation Conformance Statement (ICS) proforma in Annex A.

3 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] ISO/IEC 11571:1998, *Information technology — Telecommunications and information exchange between systems — Private Integrated Services Networks — Addressing*

[2] ISO/IEC 11572:2000, *Information technology — Telecommunications and information exchange between systems — Private Integrated Services Network — Circuit mode bearer services — Inter-exchange signalling procedures and protocol*

[3] ISO/IEC 11582:2002, *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*

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

[5] RFC 793, *Transmission Control Protocol, STD 7*, J. Postel

[6] RFC 768, *User Datagram Protocol, STD 6*, J. Postel

[7] RFC 2246, *The TLS Protocol Version 1.0*, T. Dierks and C. Allen

[8] RFC 2327, *SDP: Session Description Protocol*, M. Handley and V. Jacobson

[9] RFC 2960, *Stream Control Transmission Protocol*, R. Stewart, Q. Xie, K. Morneault, C. Sharp, H. Schwarzbauer, T. Taylor, I. Rytina, M. Kalla, L. Zhang, and V. Paxson

[10] RFC 3261, *SIP: Session Initiation Protocol*, J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, and E. Schooler

[11] RFC 3262, *Reliability of Provisional Responses in the Session Initiation Protocol (SIP)*, J. Rosenberg and H. Schulzrinne

[12] RFC 3264, *An Offer/Answer Model with the Session Description Protocol (SDP)*, J. Rosenberg and H. Schulzrinne

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

[14] RFC 3325, *Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks*, C. Jennings, J. Peterson, and M. Watson

[15] RFC 791, *Internet Protocol, STD 5*, J. Postel

[16] RFC 2460, *Internet Protocol, Version 6 (IPv6) Specification*, S. Deering and R. Hinden

[17] ITU-T Recommendation E.164, *The international public telecommunication numbering plan"*

[18] RFC 3578, *Mapping of Integrated Services Digital Network (ISDN), User Part (ISUP), Overlap Signalling to the Session Initiation Protocol (SIP)*, G. Camarillo, A. Roach, J. Peterson, and L. Ong

[19] RFC 3311, *The Session Initiation Protocol (SIP) UPDATE Method*, J. Rosenberg

[20] RFC 3420, *Internet Media Type message/sipfrag*, R. Sparks

4 Terms and definitions

For the purposes of this document, the terms and definitions given in ISO/IEC 11572, RFC 3261 and the following apply.

4.1
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 A CN can comprise a PISN, a private IP network (intranet), or a combination of the two.

4.2
gateway
entity that performs interworking between a PISN using QSIG and an IP network using SIP

4.3
IP network
network (unless otherwise stated, a corporate network) offering connectionless packet-mode services based on the Internet Protocol (IP) as the network-layer protocol

4.4
media stream
audio or other user information transmitted in UDP packets, typically containing RTP, in a single direction between the gateway and a peer entity participating in a session established using SIP

NOTE Normally a SIP session establishes a pair of media streams, one in each direction.

4.5
private integrated services network
PISN
CN or part of a CN that employs circuit-switched technology

4.6
private integrated services network exchange
PINX
PISN nodal entity comprising switching and call handling functions and supporting QSIG signalling in accordance with ISO/IEC 11572:2000

5 Abbreviated terms

DNS	Domain Name Service
IP	Internet Protocol
PINX	Private Integrated services Network eXchange
PISN	Private Integrated Services Network
RTP	Real-time Transport Protocol
SCTP	Stream Control Transmission Protocol

SDP	Session Description Protocol
SIP	Session Initiation Protocol
TCP	Transmission Control Protocol
TLS	Transport Layer Security
TU	Transaction User
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol

6 Background and Architecture

During the 1980s, corporate voice telecommunications adopted technology similar in principle to Integrated Services Digital Networks (ISDN). Digital circuit switches, commonly known as Private Branch eXchanges (PBX) or more formally as Private Integrated services Network eXchanges (PINX) have been interconnected by digital transmission systems to form Private Integrated Services Networks (PISN). These digital transmission systems carry voice or other payload in fixed-rate channels, typically 64 Kbit/s, and signalling in a separate channel. A technique known as common channel signalling is employed, whereby a single signalling channel potentially controls a number of payload channels or bearer channels. A typical arrangement is a point-to-point transmission facility at T1 or E1 rate providing a 64 Kbit/s signalling channel and 23 or 30 bearer channels, respectively. Other arrangements are possible and have been deployed, including the use of multiple transmission facilities for a signalling channel and its logically associated bearer channels. Also, arrangements involving bearer channels at sub-64 Kbit/s have been deployed, where voice payload requires the use of codecs that perform compression.

QSIG is the internationally-standardized message-based signalling protocol for use in networks as described above. It runs in a signalling channel between two PINXs and controls calls on a number of logically associated bearer channels between the same two PINXs. The signalling channel and its logically associated bearer channels are collectively known as an inter-PINX link. QSIG is independent of the type of transmission capabilities over which the signalling channel and bearer channels are provided. QSIG is also independent of the transport protocol used to transport QSIG messages reliably over the signalling channel.

QSIG provides a means for establishing and clearing calls that originate and terminate on different PINXs. A call can be routed over a single inter-PINX link connecting the originating and terminating PINX, or over several inter-PINX links in series with switching at intermediate PINXs known as transit PINXs. A call can originate or terminate in another network, in which case it enters or leaves the PISN environment through a gateway PINX. Parties are identified by numbers, in accordance with either [17] or a private numbering plan. This basic call capability is specified in [2]. In addition to basic call capability, QSIG specifies a number of further capabilities supporting the use of supplementary services in PISNs.

More recently, corporate telecommunications networks have started to exploit IP in various ways. One way is to migrate part of the network to IP using SIP. This might, for example, be a new branch office with a SIP proxy and SIP endpoints instead of a PINX. Alternatively, SIP equipment might be used to replace an existing PINX or PINXs. The new SIP environment needs to interwork with the QSIG-based PISN in order to support calls originating in one environment and terminating in the other. Interworking is achieved through a gateway.

Interworking between QSIG and SIP at gateways can also be used where a SIP network interconnects different parts of a PISN, thereby allowing calls between the different parts. A call can enter the SIP network at one gateway and leave at another. Each gateway would behave in accordance with this International Standard.

Another way of connecting two parts of a PISN would be to encapsulate QSIG signalling in SIP messages for calls between the two parts. This is outside the scope of this International Standard but could be the subject of future work.

This International Standard specifies signalling protocol interworking aspects of a gateway between a PISN employing QSIG signalling and an IP network employing SIP signalling. The gateway appears as a PINX to other PINXs in the PISN. The gateway appears as a SIP endpoint to other SIP entities in the IP network. The environment is shown in Figure 1.

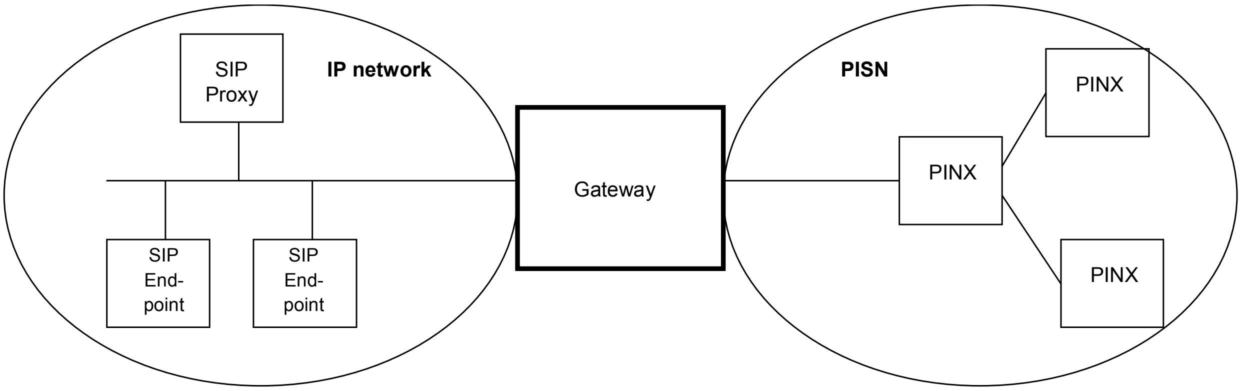

Figure 1 — Environment

In addition to the signalling interworking functionality specified in this International Standard, it is assumed that the gateway also includes the following functionality:

— one or more physical interfaces on the PISN side supporting one or more inter-PINX links, each link providing one or more constant bit rate channels for media streams and a reliable layer 2 connection (e.g., over a fixed rate physical channel) for transporting QSIG signalling messages; and

— one or more physical interfaces on the IP network side supporting, through layer 1 and layer 2 protocols, IP as the network layer protocol and UDP [6] and TCP [5] as transport layer protocols, these being used for the transport of SIP signalling messages and, in the case of UDP, also for media streams;

— optionally the support of TLS [7] and/or SCTP [9] as additional transport layer protocols on the IP network side, these being used for the transport of SIP signalling messages; and

— a means of transferring media streams in each direction between the PISN and the IP network, including as a minimum packetization of media streams sent to the IP network and de-packetization of media streams received from the IP network.

NOTE: [10] mandates support for both UDP and TCP for the transport of SIP messages and allows optional support for TLS and/or SCTP for this same purpose.

The protocol model relevant to signalling interworking functionality of a gateway is shown in Figure 2.

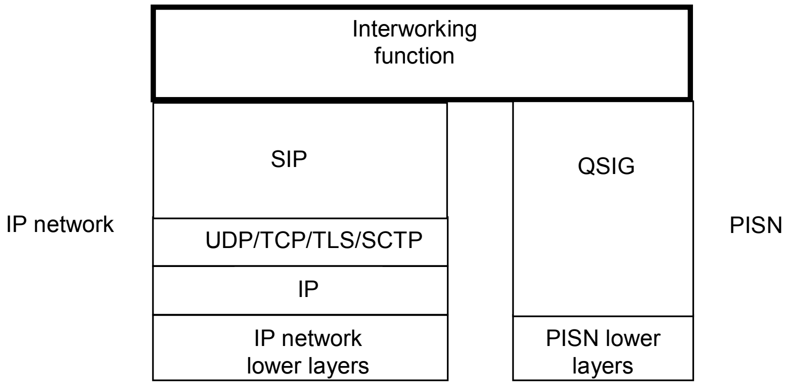

Figure 2 — Protocol model

In Figure 2, the SIP box represents SIP syntax and encoding, the SIP transport layer, and the SIP transaction layer. The Interworking function includes SIP Transaction User (TU) functionality.

The gateway maps received QSIG messages, where appropriate, to SIP messages and vice versa and maintains an association between a QSIG call and a SIP dialog.

A call from QSIG to SIP is initiated when a QSIG SETUP message arrives at the gateway. The QSIG SETUP message initiates QSIG call establishment, and an initial response message (e.g., CALL PROCEEDING) completes negotiation of the bearer channel to be used for that call. The gateway then sends a SIP INVITE request, having translated the QSIG called party number to a URI suitable for inclusion in the Request-URI. The SIP INVITE request and the resulting SIP dialog, if successfully established, are associated with the QSIG call. The SIP 2xx response to the INVITE request is mapped to a QSIG CONNECT message, signifying answer of the call. During establishment, media streams established by SIP and SDP are connected to the bearer channel.

A call from SIP to QSIG is initiated when a SIP INVITE request arrives at the gateway. The gateway sends a QSIG SETUP message to initiate QSIG call establishment, having translated the SIP Request- URI to a number suitable for use as the QSIG called party number. The resulting QSIG call is associated with the SIP INVITE request and with the eventual SIP dialog. Receipt of an initial QSIG response message completes negotiation of the bearer channel to be used, allowing media streams established by SIP and SDP to be connected to that bearer channel. The QSIG CONNECT message is mapped to a SIP 200 OK response to the INVITE request.

Annex B gives examples of typical message sequences that can arise.

7 General Requirements

In order to conform to this International Standard, a gateway SHALL support QSIG in accordance with [2] as a gateway and SHALL support SIP in accordance with [10] as a UA. In particular, the gateway SHALL support SIP syntax and encoding, the SIP transport layer, and the SIP transaction layer in accordance with [10]. In addition, the gateway SHALL support SIP TU behaviour for a UA in accordance with [10] except where stated otherwise in 8, 9, and 10 of this International Standard.

NOTE 1 [10] mandates that a SIP entity support both UDP and TCP as transport layer protocols for SIP messages. Other transport layer protocols can also be supported.

The gateway SHALL also support SIP reliable provisional responses in accordance with [11] as a UA.

NOTE 2 [11] makes provision for recovering from loss of provisional responses (other than 100) to INVITE requests when using unreliable transport services in the IP network. This is important for ensuring delivery of responses that map to essential QSIG messages.

The gateway SHALL support SDP in accordance with [8] and its use in accordance with the offer/answer model in [12].

Clause 9 also specifies optional use of the Privacy header in accordance with [13] and the P-Asserted-Identity header in accordance with [14].

The gateway SHALL support calls from QSIG to SIP and calls from SIP to QSIG.

SIP methods not defined in [10] or [11] are outside the scope of this International Standard but could be the subject of other standards for interworking with QSIG, e.g., for interworking in support of supplementary services.

As a result of DNS lookup by the gateway in order to determine where to send a SIP INVITE request, a number of candidate destinations can be attempted in sequence. The way in which this is handled by the gateway is outside the scope of this International Standard. However, any behaviour specified in this International Standard on receipt of a SIP 4xx or 5xx final response to an INVITE request SHOULD apply only when there are no more candidate destinations to try or when overlap signalling applies in the SIP network (see 8.2.2.2).

8 Message Mapping Requirements

8.1 Message Validation and Handling of Protocol Errors

The gateway SHALL validate received QSIG messages in accordance with the requirements of [2] and SHALL act in accordance with [2] on detection of a QSIG protocol error. The requirements of this Clause for acting on a received QSIG message apply only to a received QSIG message that has been successfully validated and that satisfies one of the following conditions:

— the QSIG message is a SETUP message and indicates a destination in the IP network and a bearer capability for which the gateway is able to provide interworking; or

— the QSIG message is a message other than SETUP and contains a call reference that identifies an existing call for which the gateway is providing interworking between QSIG and SIP.

The processing of any valid QSIG message that does not satisfy any of these conditions is outside the scope of this International Standard. Also, the processing of any QSIG message relating to call-independent signalling connections or connectionless transport, as specified in [3], is outside the scope of this International Standard.

If segmented QSIG messages are received, the gateway SHALL await receipt of all segments of a message and SHALL validate and act on the complete reassembled message.

The gateway SHALL validate received SIP messages (requests and responses) in accordance with the requirements of [10] and SHALL act in accordance with [10] on detection of a SIP protocol error.

Requirements of this Clause for acting on a received SIP message apply only to a received message that has been successfully validated and that satisfies one of the following conditions:

— the SIP message is an INVITE request that contains no tag parameter in the To header field, does not match an ongoing transaction (i.e., is not a merged request; see 8.2.2.2 of [10]), and indicates a destination in the PISN for which the gateway is able to provide interworking; or

— the SIP message is a request that relates to an existing dialog representing a call for which the gateway is providing interworking between QSIG and SIP; or

— the SIP message is a CANCEL request that relates to a received INVITE request for which the gateway is providing interworking with QSIG but for which the only response sent is informational (1xx), no dialog having been confirmed; or

— the SIP message is a response to a request sent by the gateway in accordance with this Clause.

The processing of any valid SIP message that does not satisfy any of these conditions is outside the scope of this International Standard.

NOTE These rules mean that an error detected in a received message will not be propagated to the other side of the gateway. However, there can be an indirect impact on the other side of the gateway, e.g., the initiation of call clearing procedures.

The gateway SHALL run QSIG protocol timers as specified in [2] and SHALL act in accordance with [2] if a QSIG protocol timer expires. Any other action on expiry of a QSIG protocol timer is outside the scope of this International Standard, except that if it results in the clearing of the QSIG call, the gateway SHALL also clear the SIP call in accordance with 8.4.5.

The gateway SHALL run SIP protocol timers as specified in [10] and SHALL act in accordance with [10] if a SIP protocol timer expires. Any other action on expiry of a SIP protocol timer is outside the scope of this International Standard, except that if it results in the clearing of the SIP call, the gateway SHALL also clear the QSIG call in accordance with 8.4.5.

8.2 Call Establishment from QSIG to SIP

8.2.1 Call Establishment from QSIG to SIP Using En Bloc Procedures

The following procedures apply when the gateway receives a QSIG SETUP message containing a Sending Complete information element or the gateway receives a QSIG SETUP message and is able to determine that the number in the Called party number information element is complete.

NOTE In the absence of a Sending Complete information element, the means by which the gateway determines the number to be complete is an implementation matter. It can involve knowledge of the numbering plan and/or use of inter-digit timer expiry.

8.2.1.1 Receipt of QSIG SETUP Message

On receipt of a QSIG SETUP message containing a number that the gateway determines to be complete in the Called party number information element, or containing a Sending complete information element and a number that could potentially be complete, the gateway SHALL map the QSIG SETUP message to a SIP INVITE request. The gateway SHALL also send a QSIG CALL PROCEEDING message.

The gateway SHALL generate the SIP Request-URI, To, and From fields in the SIP INVITE request in accordance with Clause 9. The gateway SHALL include in the INVITE request a Supported header containing option tag 100rel, to indicate support for [11].

The gateway SHALL include SDP offer information in the SIP INVITE request as described in Clause 10. It SHOULD also connect the incoming media stream to the user information channel of the inter- PINX link, to allow the caller to hear in-band tones or announcements and prevent speech clipping on answer. Because of forking, the gateway may receive more than one media stream, in which case it SHOULD select one (e.g., the first received). If the gateway is able to correlate an unselected media stream with a particular early dialog established using a reliable provisional response, it MAY use the UPDATE method [19] to stop that stream and then use the UPDATE method to start that stream again if a 2xx response is received on that dialog.

On receipt of a QSIG SETUP message containing a Sending complete information element and a number that the gateway determines to be incomplete in the Called party number information element, the gateway SHALL initiate QSIG call clearing procedures using cause value 28, "invalid number format (address incomplete)".

If information in the QSIG SETUP message is unsuitable for generating any of the mandatory fields in a SIP INVITE request (e.g., if a Request-URI cannot be derived from the QSIG Called party number information element) or for generating SDP information, the gateway SHALL NOT issue a SIP INVITE request and SHALL initiate QSIG call clearing procedures in accordance with [2].

8.2.1.2 Receipt of SIP 100 (Trying) Response to an INVITE Request

A SIP 100 response SHALL NOT trigger any QSIG messages. It only serves the purpose of suppressing INVITE request retransmissions.

8.2.1.3 Receipt of SIP 18x provisional response to an INVITE request

The gateway SHALL map a received SIP 18x response to an INVITE request to a QSIG PROGRESS or ALERTING message based on the following conditions.

— If a SIP 180 response is received and no QSIG ALERTING message has been sent, the gateway SHALL generate a QSIG ALERTING message. The gateway MAY supply ring-back tone on the user information channel of the inter-PINX link, in which case the gateway SHALL include progress description number 8 in the QSIG ALERTING message. Otherwise the gateway SHALL NOT include progress description number 8 in the QSIG ALERTING message unless the gateway is aware that in-band information (e.g., ring-back tone) is being transmitted.

— If a SIP 181/182/183 response is received, no QSIG ALERTING message has been sent, and no message containing progress description number 1 has been sent, the gateway SHALL generate a QSIG PROGRESS message containing progress description number 1.

NOTE This will ensure that QSIG timer T310 is stopped if running at the Originating PINX.

In all other scenarios, the gateway SHALL NOT map the SIP 18x response to a QSIG message.

If the SIP 18x response contains a Require header with option tag 100rel, the gateway SHALL send back a SIP PRACK request in accordance with [44].

8.2.1.4 Receipt of SIP 2xx Response to an INVITE Request

If the gateway receives a SIP 2xx response as the first SIP 2xx response to a SIP INVITE request, the gateway SHALL map the SIP 2xx response to a QSIG CONNECT message. The gateway SHALL also send a SIP ACK request to acknowledge the 2xx response. The gateway SHALL NOT include any SDP information in the SIP ACK request. If the gateway receives further 2xx responses, it SHALL respond to each in accordance with [10], SHOULD issue a BYE request for each, and SHALL NOT generate any further QSIG messages.

Media streams will normally have been established in the IP network in each direction. If so, the gateway SHALL connect the media streams to the corresponding user-information channel on the inter- PINX link if it has not already done so and stop any local ring-back tone.

If the SIP 2xx response is received in response to the SIP PRACK request, the gateway SHALL NOT map this message to any QSIG message.

NOTE A SIP 2xx response to the INVITE request can be received later on a different dialog as a result of a forking proxy.

8.2.1.5 Receipt of SIP 3xx Response to an INVITE Request

On receipt of a SIP 3xx response to an INVITE request, the gateway SHALL act in accordance with [10].

NOTE This will normally result in sending a new SIP INVITE request.