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

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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 17343 was prepared by Joint Technical Committee ISO/IEC JTC 1, *Information technology*, Subcommittee SC 6, *Telecommunications and information exchange between systems*.

ISO/IEC 17343 was prepared by ECMA (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.

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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.

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

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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).

"QSIG" is a signalling protocol that operates between Private Integrated Services 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 ISO/IEC 11572 (call control in support of basic services), ISO/IEC 11582 (generic functional protocol for the support of supplementary services) and a number of Standards specifying individual supplementary services.

SIP is an application layer protocol for establishing, terminating and modifying multimedia sessions. It is typically carried over the Internet Protocol (IP) (IETF RFC 791 and IETF RFC 2460). Telephone calls are considered as a type of multimedia session where just audio is exchanged. SIP is defined in IETF RFC 3261.

This International Standard specifies 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. 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.

A brief assessment of security considerations in the IP network resulting from the interworking specified in this International Standard is given in annex C.

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.

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

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*

ISO/IEC 11579-1:1994, *Information technology — Telecommunications and information exchange between systems — Private Integrated Services Network — Part 1: Reference configuration for PISN Exchanges (PINX)*

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*

ISO/IEC 21409:2001, *Information technology — Telecommunications and information exchange between systems — Corporate telecommunication networks — Signalling interworking between QSIG and H.323 — Generic functional protocol for the support of supplementary services*

ITU-T Rec. E.164:1997, *The International Public Telecommunication Numbering Plan*

IETF RFC 768, *"User Datagram Protocol"; J. Postel*

IETF RFC 791, *"Internet Protocol"; J. Postel*

IETF RFC 793, *"Transmission Control Protocol"; J. Postel*

IETF RFC 2046, *"Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types"; N. Freed/ N. Borenstein*

IETF RFC 2246, *"The TLS protocol version 1.0", T. Dierks, C. Allen*

IETF RFC 2327, *"SDP: Session Description Protocol"; M. Handley/V. Jacobson*

IETF RFC 2460, *"Internet Protocol, Version 6 (IPv6)"; S. Deering, R. Hinden*

IETF RFC 2960, *"Stream Control Transmission Protocol"; R. Stewart et al.*

IETF RFC 3261, *"SIP: Session Initiation Protocol"; M. Handley/H. Schulzrinne/E. Schooler/J. Rosenberg*

IETF RFC 3262, *"Reliability of Provisional Responses in SIP"; J. Rosenberg /H. Schulzrinne*

IETF RFC 3264, *"An Offer/Answer Model with SDP"; J. Rosenberg /H. Schulzrinne*

IETF RFC 3323, *"A Privacy Mechanism for the Session Initiation Protocol (SIP)"; J. Peterson*

IETF RFC 3325, *"Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks"; C. Jennings, J. Peterson, M. Watson*

4   Definitions

For the purposes of this International Standard, the following definitions apply.

4.1   External definitions

This International Standard uses the following terms defined in other documents:

- Call   (ISO/IEC 21409)

- Corporate telecommunication network (CN)   (ISO/IEC 21409)

- Private Integrated Services Network (PISN)   (ISO/IEC 21409)

- Private Integrated services Network eXchange (PINX)   (ISO/IEC 11579-1)

Additionally the definitions in ISO/IEC 11572 and IETF RFC 3261 apply as appropriate.

4.2   Other definitions

4.2.1
Gateway
An entity that performs interworking between a PISN using QSIG and an IP network using SIP.

4.2.2
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.

4.2.3
Media stream
Audio or other user information transmitted in UDP packets 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.

5   Acronyms

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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 24 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 ITU-T recommendation E.164 or a private numbering plan. This basic call capability is specified in ISO/IEC 11572. 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.

Another way of migrating is to use a SIP network to interconnect two parts of a PISN and encapsulate QSIG signalling in SIP messages for calls between the two parts of the PISN. This is outside the scope of this specification 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.
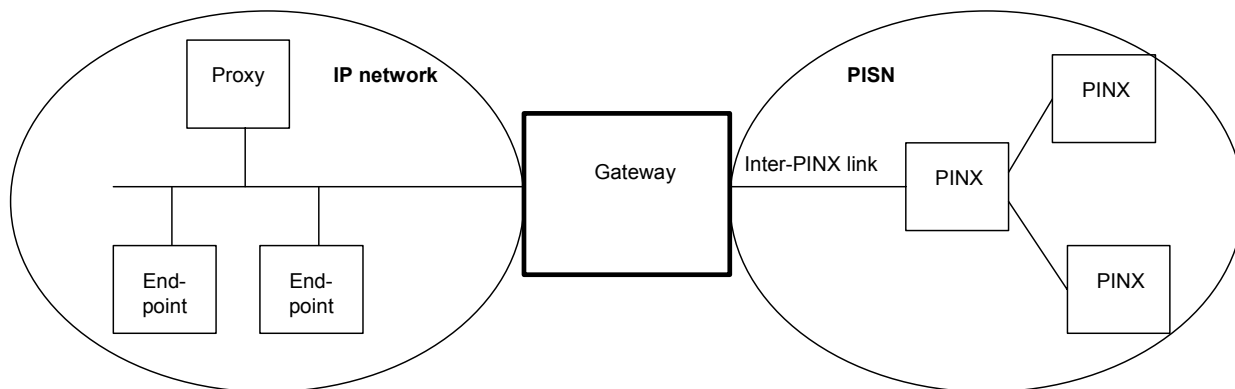

Figure 1 — Environment

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 information 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 (IETF RFC 768) and TCP (IETF RFC 793) as transport layer protocols, these being used for the transport of SIP signalling messages and, in the case of UDP, also for media information;

- optionally the support of TLS (IETF RFC 2246) and/or SCTP (IETF RFC 2960) 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 information in each direction between the PISN and the IP network, including as a minimum packetization of media information sent to the IP network and de-packetization of media information received from the IP network.

NOTE   IETF RFC 3261 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.
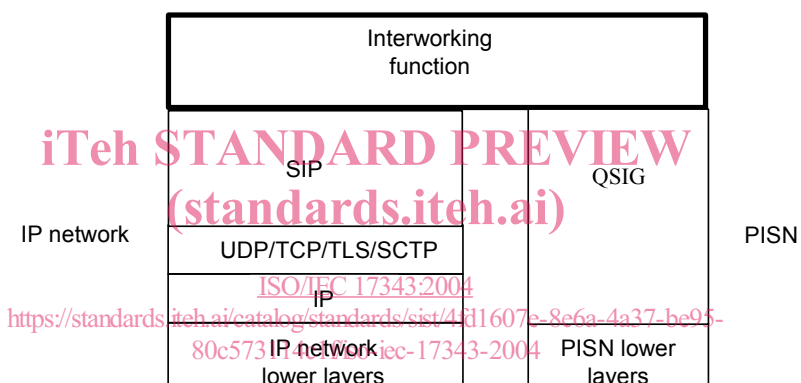
Figure 2 — Protocol model

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 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 200 OK response 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.

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 ISO/IEC 11572 as a gateway and shall support SIP in accordance with IETF RFC 3261 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 IETF RFC 3261. In addition, the gateway shall support SIP TU behaviour for a UA in accordance with IETF RFC 3261 except where stated otherwise in this International Standard.

NOTE 1   IETF RFC 3261 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 IETF RFC 3262 as a UA.

NOTE 2   IETF RFC 3262 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 IETF RFC 2327 and its use in accordance with the offer / answer model in IETF RFC 3264.

Clause 9 also specifies optional use of the Privacy header in accordance with IETF RFC 3323 and the P-Asserted-Identity header in accordance with IETF RFC 3325.

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

SIP methods not defined in IETF RFC 3261, IETF RFC 3262, IETF RFC 3323 or IETF RFC 3325 are outside the scope of this International Standard but could be the subject of other specifications for interworking with QSIG, e.g., for interworking in support of supplementary services.

As a result of DNS look-up 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 final response should apply only when there are no more candidate destinations to try.

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 ISO/IEC 11572 and shall act in accordance with ISO/IEC 11572 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.

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 IETF RFC 3261 and shall act in accordance with IETF RFC 3261 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 IETF RFC 3261) 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 ISO/IEC 11572 and shall act in accordance with ISO/IEC 11572 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 IETF RFC 3261 and shall act in accordance with IETF RFC 3261 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 enbloc 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   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 the gateway cannot determine to 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 IETF RFC 3262.

The gateway shall include SDP information in the SIP INVITE request as described in clause 10.

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 ISO/IEC 11572.

8.2.1.2 Receipt of SIP 100 (Trying) response

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

The gateway shall map a received SIP 18x response 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 a media stream has been established towards the gateway and 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, no QSIG PROGRESS message containing progress description number 8 has been sent and a media stream has been established towards the gateway, the gateway shall generate a QSIG PROGRESS message. The QSIG PROGRESS message shall contain progress description number 8 in a Progress indicator information element. The gateway shall also connect the media streams to the corresponding user information channel of the inter-PINX link.

- If a SIP 181/182/183 response is received, no QSIG ALERTING message has been sent, no QSIG PROGRESS message containing progress description number 1 or 8 has been sent and no media stream has been established towards the gateway, the gateway shall generate a QSIG PROGRESS message. The QSIG PROGRESS message shall contain progress description number 1 in a Progress indicator information element.

NOTE 1   This will ensure that QSIG timer T310 is stopped if running at the Originating PINX.

NOTE 2   Media streams are established as a result of receiving SDP answer information in a reliable provisional response and can be modified by means of the SIP UPDATE method (IETF RFC 3311). If a media stream is established towards the gateway, connecting the media stream to the corresponding user information channel on the inter-PINX link will allow the caller to hear in-band tones or announcements.

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 IETF RFC 3262.

8.2.1.4 Receipt of SIP 2xx response

If the gateway receives a SIP 200 (OK) response as the first SIP 200 response to a SIP INVITE request, the gateway shall map the SIP 200 (OK) response to a QSIG CONNECT message. The gateway shall also send a SIP ACK request to acknowledge the 200 (OK) response. The gateway shall not include any SDP information in the SIP ACK request. If the gateway receives further 200 (OK) responses, it shall respond to each in accordance with IETF RFC 3261 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 200 (OK) response is received in response to the SIP PRACK request, the gateway shall not map this message to any QSIG message.

If the gateway receives a SIP 2xx response other than 200 (OK), the gateway shall send a SIP ACK request.

NOTE   A SIP 200 (OK) response can be received later as a result of a forking proxy.

8.2.1.5   Receipt of SIP 3xx response

On receipt of a SIP 3xx response, the gateway shall act in accordance with IETF RFC 3261.

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

Unless the gateway supports the QSIG Call Diversion Supplementary Service, no QSIG message shall be sent. The definition of Call Diversion Supplementary Service for QSIG to SIP interworking is beyond the scope of this International Standard.

8.2.2   Call establishment from QSIG to SIP using overlap procedures

SIP uses en-bloc signalling and it is **strongly recommended to avoid using overlap signalling in a SIP network**. A SIP/QSIG gateway dealing with overlap signalling, should perform a conversion from overlap to en-bloc signalling method using one or more of the following mechanisms:

• timers;

• numbering plan information;

• the presence of a Sending complete information element in a received QSIG INFORMATION message.

If the gateway performs a conversion from overlap to en-bloc signalling in the SIP network then the procedures defined in 8.2.2.1 shall apply.

However, for some applications it might be impossible to avoid using overlap signalling in the SIP network. In this case the procedures defined in 8.2.2.2 shall apply.

8.2.2.1   Enbloc signalling in SIP network

8.2.2.1.1   Receipt of QSIG SETUP message

On receipt of a QSIG SETUP message containing no Sending complete information element and a number in the Called party number information element that the gateway cannot determine to be complete, the gateway shall send back a QSIG SETUP ACKNOWLEDGE message, start QSIG timer T302 and await further number digits.

8.2.2.1.2   Receipt of QSIG INFORMATION message

On receipt of each QSIG INFORMATION message containing no Sending complete information element and containing a number that the gateway cannot determine to be complete, QSIG timer T302 shall be restarted. When QSIG timer T302 expires or a QSIG INFORMATION message containing a Sending complete information element is received the gateway shall send a SIP INVITE request as described in 8.2.1.1. The Request-URI and To fields (see clause 9) shall be generated from the concatenation of information in the Called party number information element in the received QSIG SETUP and INFORMATION messages. The gateway shall also send a QSIG CALL PROCEEDING message.