Internet Engineering Task Force J. Elwell
Internet Draft Siemens
F. Derks
Philips
P. Mourot/O. Rousseau
Alcatel
draft-elwell-sipping-qsig2sip-00.txt
Expires: October 2002 April 2002

Interworking between SIP and QSIG

Status of this Memo

Abstract

This document specifies interworking between the Session Initiation Protocol (SIP) and QSIG within corporate 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. 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 ECMA Standards and published also as ISO/IEC standards.

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 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 document is a product of the authors' activities in ECMA (www.ecma.ch) on interoperability of QSIG with IP networks.

1 Introduction

This document 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 at the Q reference point between Private Integrated Services eXchanges (PINX) within a Private Integrated Services Network (PISN). The Q reference point is defined in ECMA-133. A PISN provides circuit-switched basic services and supplementary services to its users. QSIG is specified in ECMA Standards, in particular ECMA-143 (call control in support of basic services), ECMA-165 (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 IP. Telephone calls are considered as a type of multimedia session where just audio is exchanged. SIP is defined in IETF RFC 3261.

This document 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 specification.

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 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 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 [2] and indicate requirement levels for compliant SIP implementations.

3 Definitions

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

3.1 External definitions

This specification uses the following terms defined in other documents:

-Call (ECMA-307)
-Corporate telecommunication network (CN) (ECMA-307)
-Private Integrated Services Network (PISN) (ECMA-307)
-Private Integrated services Network eXchange (PINX) (ECMA-133)

Additionally the definitions in ECMA-143 and IETF RFC 3261 apply as appropriate.

3.2 Other definitions

3.2.1 Gateway

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

3.2.2 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.

3.2.3 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 Acronyms

DNS	Domain Name Service
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

5 Architecture

This document 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.

```
       +------+   IP network                    PISN
       |      |
       |SIP   |                                               +------+
       |Proxy |                                              /|      |
       |      |                                             / |PINX  |
       +---+--+               *-----------+                /  |      |
           |                  |           |        +-----+/   +------+
           |                  |           |        |     |
           |                  |           |        |PINX |
-----+-----+---------+--------+  Gateway  +---------|     |
     |               |        |           |        |     |\
     |               |        |           |        +-----+ \
     |               |        |           |                 \ +------+
     |               |        |           |                  \|      |
  +--+---+        +--+---+    *-----------+                   |PINX  |
  |SIP   |        |SIP   |                                    |      |
  |End-  |        |End-  |                                    +------+
  |point |        |point |
  +------+        +------+
```

Figure 1 – Environment

In addition to the signalling interworking functionality specified in this specification, 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 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 (RFC 768) and TCP (RFC 761) 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 (RFC 2246) and/or SCTP (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. 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.

```
+-----------------------------------------------------------+
|                  Inter-working function                   |
|                                                           |
+-----------------------+----------+------------------------+
|                       |          |                        |
|         SIP           |          |                        |
|                       |          |                        |
+-----------------------+          |                        |
|                       |          |                        |
|  UDP/TCP/TLS/SCTP     |          |         QSIG           |
|                       |          |                        |
+-----------------------+          |                        |
|                       |          |                        |
|         IP            |          |                        |
|                       |          |                        |
+-----------------------+          +------------------------+
|     IP network        |          |         PISN           |
|     lower layers      |          |     lower layers       |
|                       |          |                        |
+-----------------------+          +------------------------+
```

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.

6 Overview

The gateway maps received QSIG messages, where appropriate, to SIP messages and vice versa. Annex A gives examples of typical message sequences that can arise.

7 General requirements

In order to conform to this specification, a gateway SHALL support QSIG in accordance with ECMA-143 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 RFC 3261. In addition, the gateway SHALL support SIP TU behaviour for a UA in accordance with RFC 3261 except where stated otherwise in this specification.

NOTE 1. 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 BBBB as a UA.

NOTE 2. RFC BBBB 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 RFC 2327 and its use in accordance with the offer / answer model in RFC CCCC.

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

SIP methods not defined in RFC 3261 or RFC BBBB are outside the scope of this specification 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 specification. However, any behaviour specified in this document on receipt of a SIP final response SHOULD apply only when a final response is received and 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 ECMA-143 and SHALL act in accordance with ECMA-143 on detection of a QSIG protocol error. The requirements of this section 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 specification.

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 section 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 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 section.

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

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.

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 section 9. The gateway SHALL include in the INVITE request a Supported header containing option tag 100rel, to indicate support for RFC BBBB.

The gateway SHALL include SDP information in the SIP INVITE request as described in section 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 ECMA-143.

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 QSIG ALERTING message SHALL contain a Progress indicator information element containing progress description number 8. If the SDP answer has been received, the gateway SHALL connect the media streams to the corresponding user information channel of the inter-PINX link. If the SDP answer has not been received, the gateway SHALL supply ring-back tone on the user information channel of the inter-PINX link. If the

SDP answer is subsequently received, the gateway SHALL stop ring-back tone and 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 8 has been sent and the SDP answer has been received, 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 the SDP answer has not been received, 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. 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.

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 RFC 3261 and SHALL NOT generate any further QSIG messages.

The SDP answer will now have been received. 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.

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 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 specification.

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 T302 timer 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, timer T302 SHALL be restarted. When 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 section 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.

8.2.2.1.3 Receipt of SIP responses

SIP responses SHALL be mapped as described in 8.2.1.

8.2.2.2 Overlap signalling in SIP network

8.2.2.2.1 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 and start QSIG timer T302. If the QSIG SETUP message contains the minimum number of digits required to route the call in the IP network, the gateway SHALL send a SIP INVITE request as specified in 8.2.1.1. Otherwise the gateway SHALL wait for more digits to arrive in QSIG INFORMATION messages.

8.2.2.2.2 Receipt of QSIG INFORMATION message

On receipt of a QSIG INFORMATION message the gateway SHALL restart the QSIG T302 timer. Further behaviour of the gateway SHALL depend on whether or not it has already sent a SIP INVITE request. If the gateway has not sent a SIP INVITE request and it now has the minimum number of digits required to route the call, it SHALL send a SIP INVITE request as specified in 8.2.2.1.2. If the gateway still does not have the minimum number of digits required than it SHALL wait for more QSIG INFORMATION messages to arrive.

If the gateway has already sent one or more SIP INVITE requests, and whether or not final responses to those requests have been received, it SHALL send a new SIP INVITE request with the new digits. The new SIP INVITE request SHALL have the same Call-ID as the first SIP INVITE request sent but SHALL have updated Request-URI and To fields. The updated Request-URI and To fields (see section 9) SHALL be generated from the concatenation of information in the Called party number information element in the received QSIG SETUP and INFORMATION messages.

NOTE. The first SIP INVITE request and all subsequent SIP INVITE requests sent in this way belong to the same call but to different dialogs.

8.2.2.2.3 Receipt of SIP 100 (Trying) response

The requirements of 8.2.1.2 SHALL apply.

8.2.2.2.4 Receipt of SIP 18x provisional response

The requirements of 8.2.1.3 SHALL apply.

8.2.2.2.5 Receipt of SIP 2xx response

The requirements of 8.2.1.4 SHALL apply.

8.2.2.2.6 Receipt of SIP 3xx response

The requirements of 8.2.1.5 SHALL apply.

8.2.2.2.7 Receipt of a SIP 484 (Address Incomplete) response

The SIP 484 response indicates that more digits are required to complete the call. On receipt of a SIP 484 response the gateway SHALL send back a SIP ACK request. The gateway SHALL also send a QSIG DISCONNECT message if either of the following conditions apply:

-T302 expires and all the SIP INVITE requests sent have been answered with a final response other than 200 OK; or

-a QSIG INFORMATION message containing a Sending complete information element has been received and all the SIP INVITE requests sent have been answered with a final response (other than 200 OK).

In all other cases the receipt of a SIP 484 response SHALL NOT trigger the sending of any QSIG message.

8.2.2.2.8 Receipt of a SIP 4xx (except 484), 5xx or 6xx response

If a SIP 4xx (except 484), 5xx or 6xx final response arrives for a pending SIP INVITE transaction, the gateway SHALL send a SIP ACK request. If this occurs before T302 has expired, the gateway shall either send a QSIG DISCONNECT message (8.4.4) or behave as for a SIP 484 response (8.2.2.2.7).

8.2.2.2.9 Receipt of multiple SIP responses

The responses to all the SIP INVITE requests sent except for the last one are typically SIP 4xx responses (e.g. 484 (Address Incomplete)) that terminate the SIP INVITE transaction.

However, the gateway can receive a SIP 183 (Session Progress) response with a media description. The media stream will typically contain a message such as "…We are trying to connect you… ". The issue of receiving different SIP 183 (Session Progress) responses with media descriptions for different SIP INVITE transactions is a gateway concern. The gateway SHOULD decide which media stream (if any) are to be played to the user.

8.2.2.2.10 Cancelling pending SIP INVITE transactions

When a gateway sends a new SIP INVITE request containing new digits, it SHOULD NOT send a SIP CANCEL request to cancel the previous SIP INVITE transaction. This SIP CANCEL request could arrive at an egress gateway before the new SIP INVITE request and trigger premature call clearing.

NOTE. Previous SIP INVITE transactions can be expected to result in SIP 4xx class responses, which terminate the transaction.

8.2.2.2.11 SIP INVITE requests reaching multiple gateways

Each SIP INVITE request sent by a gateway represents a new transaction and hence can be routed differently. For instance, the first SIP INVITE request might be routed to a particular egress gateway and a subsequent SIP INVITE request to another gateway. The result is that both gateways initiate call establishment in the remote network. Since one of the call establishments has an incomplete destination number, it can be expected to fail, having already consumed resources in the remote network.

To avoid this problem it is RECOMMENDED that all the SIP INVITE requests should follow the same path as the first one. This would however restrict the number of services the SIP network can provide. It would not be possible to route a subsequent SIP INVITE request to an application server just because the previous one was routed in a different way.

This issue should be taken into consideration before using overlap signalling in SIP. If initiating multiple call establishments in the remote network is not acceptable in a particular application, overlap signalling SHOULD NOT be used.

8.3 Call Establishment from SIP to QSIG

8.3.1 Receipt of SIP INVITE request for a new call

On receipt of a SIP INVITE request for a new call, the gateway SHALL generate a QSIG SETUP message from the received SIP INVITE request. The gateway SHALL generate the Called party number and Calling party number information elements in accordance with section 9 and SHALL generate the Bearer capability information element in accordance with section 10. If the gateway can determine that the number placed in the Called party number information element is complete, the gateway MAY include the Sending complete information element.

NOTE 1. 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 the inter-digit timer.

The gateway SHOULD send a SIP 100 (Trying) response.

If information in the SIP INVITE request is unsuitable for generating any of the mandatory information elements in a QSIG SETUP message (e.g., if a QSIG Called party number information element cannot be derived from SIP Request-URI field), the gateway SHALL NOT issue a QSIG SETUP message and SHALL send a SIP 4xx, 5xx or 6xx response.

If the SIP INVITE request does not contain SDP information and does not contain either a Required header or a Supported header with option tag 100rel, the gateway SHALL NOT issue a QSIG SETUP message and SHALL send a SIP 488 (Not Acceptable Here) response.

NOTE 2. The absence of SDP offer information in the SIP INVITE request means that the gateway might need to send SDP offer information in a provisional response in order to ensure that tones and announcements from the PISN are transmitted. SDP offer information cannot be sent in an unreliable provisional response because the SDP answer would need to be returned in a SIP PRACK request.

On receipt of a SIP INVITE request relating to a call that has already been established from SIP to QSIG, the procedures of 8.3.9 SHALL apply.

8.3.2 Receipt of QSIG CALL PROCEEDING message

The receipt of a QSIG CALL PROCEEDING message SHALL NOT result in any SIP message being sent.

8.3.3 Receipt of QSIG PROGRESS message

A QSIG PROGRESS message can be received in the event of interworking at the egress from the PISN or if the PISN is usable to complete the call and generates an in-band tone or announcement. In the latter case a Cause information element is included in the QSIG PROGRESS message.

The gateway SHALL map a received QSIG PROGRESS message to a SIP 183 (Session Progress) response. If the SIP INVITE request contained either a Require header or a Supported header with option tag 100rel, the gateway SHALL include in the SIP 183 response a Require header with option tag 100rel.

If the QSIG PROGRESS message contained a Progress indicator information element with Progress description number 1 or 8, the gateway SHALL connect the media streams to the corresponding user information channel of the inter-PINX link if it has not already done so, provided the SDP answer is included in the transmitted SIP response or has already been sent or received. Inclusion of SDP offer or answer information in the 183 provisional response SHALL be in accordance with 8.3.5

If the QSIG PROGRESS message is received with a Cause information element, the gateway SHALL either wait until the tone/announcement is complete or has been applied for sufficient time before initiating call clearing, or wait for a SIP CANCEL request. If call clearing is initiated, the cause value in the QSIG PROGRESS message SHALL be used to derive the response to the SIP INVITE request in accordance with table 1.

8.3.4 Receipt of QSIG ALERTING message

The gateway SHALL map a QSIG ALERTING message to a SIP 180 (Ringing) response. If the SIP INVITE request contained either a Require header or a Supported header with option tag 100rel, the gateway SHALL include in the SIP 180 response a Require header with option tag 100rel.

If the QSIG ALERTING message contained a Progress indicator information element with Progress description number 1 or 8, the gateway SHALL connect the media streams to the corresponding user information channel of the inter-PINX link if it has not already done so, provided the SDP answer is included in the transmitted SIP response or has already been sent or received. Inclusion of SDP offer or answer information in the 180 provisional response SHALL be in accordance with 8.3.5

8.3.5 Inclusion of SDP information in a SIP 18x provisional response

When sending a SIP 18x provisional response, the gateway SHALL include SDP information in accordance with the following rules.

If the SIP INVITE request contained a Required or Supported header with option tag 100rel, and if SDP offer and answer have already been exchanged, no SDP SHALL be included in the SIP 18x provisional response.

If the SIP INVITE request contained a Required or Supported header with option tag 100rel, and if SDP offer was received in the SIP INVITE request but no SDP answer has been sent, SDP answer SHALL be included in the SIP 18x provisional response.

If the SIP INVITE request contained a Required or Supported header with option tag 100rel, and if no SDP offer was received in the SIP INVITE request and no SDP offer has already been sent, SDP offer SHALL be included in the SIP 18x provisional response.

NOTE 1. In this case, SDP answer can be expected in the SIP PRACK.

If the SIP INVITE request contained neither a Required nor a Supported header with option tag 100rel, SDP answer SHALL be included in the SIP 18x provisional response.

NOTE 2. Because the provisional response is unreliable, SDP answer needs to be repeated in each provisional response and in the final SIP 2xx response.

NOTE 3. If the SIP INVITE request contained no SDP offer and neither a Required nor a Supported header with option tag 100rel, it should have been rejected in accordance with 8.3.1.

8.3.6 Receipt of QSIG CONNECT message

The gateway SHALL map a QSIG CONNECT message to a SIP 200 (OK) final response for the SIP INVITE request. The gateway SHALL also send a QSIG CONNECT ACKNOWLEDGE message.

If the SIP INVITE request contained a Required or Supported header with option tag 100rel, and if SDP offer and answer have already been exchanged, no SDP SHALL be included in the SIP 200 response.

If the SIP INVITE request contained a Required or Supported header with option tag 100rel, and if SDP offer was received in the SIP INVITE request but no SDP answer has been sent, SDP answer SHALL be included in the SIP 200 response.

If the SIP INVITE request contained a Required or Supported header with option tag 100rel, and if no SDP offer was received in the SIP INVITE request and no SDP offer has already been sent, SDP offer SHALL be included in the SIP 200 response.

NOTE 1. In this case, SDP answer can be expected in the SIP ACK.

If the SIP INVITE request contained neither a Required nor a Supported header with option tag 100rel, SDP answer SHALL be included in the SIP 200 response.

NOTE 2. Because the provisional response is unreliable, SDP answer needs to be repeated in each provisional response and in the final 2xx response.

NOTE 3. If the SIP INVITE request contained no SDP offer and neither a Required nor a Supported header with option tag 100rel, it should have been rejected in accordance with 8.3.1.

The gateway SHALL connect the media streams to the corresponding user information channel of the inter-PINX link if it has not already done so, provided the SDP answer is included in the transmitted SIP response or has already been sent or received.

8.3.7 Receipt of SIP PRACK request

The receipt of a SIP PRACK request SHALL NOT result in any QSIG message being sent. The gateway SHALL send back a SIP 200 (OK) response to the SIP PRACK request.

If the SIP PRACK contains SDP answer and a QSIG message containing a Progress indicator information element with progress description number 1 or 8 has been received, the gateway SHALL connect the media streams to the corresponding user information channel of the inter-PINX link.

8.3.8 Receipt of SIP ACK request

The receipt of a SIP ACK request SHALL NOT result in any QSIG message being sent.

If the SIP ACK contains SDP answer, the gateway SHALL connect the media streams to the corresponding user information channel of the inter-PINX link if it has not already done so.

8.3.9 Receipt of a SIP INVITE request for a call already being established

For a call from SIP using overlap procedures, the gateway will receive multiple SIP INVITE requests that belong to the same call but have different Request-URI and To fields. Each SIP INVITE request belongs to a different dialog.

If a gateway receives a SIP INVITE request with the same Call-ID as an existing call for which the QSIG state is overlap sending and with updated Request-URI and To fields from which a called party number with a superset of digits can be derived, it SHALL generate a QSIG INFORMATION message using the call reference of the existing QSIG call instead of a new QSIG SETUP message. It SHALL also respond to the SIP INVITE request received previously with a SIP 484 Address Incomplete response.

If a gateway receives a SIP INVITE request with the same Call-ID as an existing call but failing to meet the other conditions above, the gateway SHALL clear the call by sending back a SIP 485 (Ambiguous) response and a QSIG DISCONNECT message with Cause Value 16 (Normal call clearing).

8.4 Call clearing

8.4.1 Receipt of a QSIG DISCONNECT, RELEASE or RELEASE COMPLETE message

On receipt of QSIG DISCONNECT, RELEASE or RELEASE COMPLETE message as the first QSIG call clearing message, gateway behaviour SHALL depend on the state of call establishment.

1)If the gateway has sent or received a SIP 200 (OK) response (indicating that call establishment is complete) and received a SIP ACK request, the gateway SHALL send a SIP BYE request to clear the call.

2)If the gateway has sent a SIP 200 (OK) response (indicating that call establishment is complete) but not received a SIP ACK request, the gateway SHALL wait until a SIP ACK is received and then send a SIP BYE request to clear the call.

3)If the gateway has sent a SIP INVITE request and received a SIP provisional response but not a SIP final response, the gateway SHALL send a SIP CANCEL request to clear the call.

NOTE. In accordance with RFC 3261, if after sending a SIP CANCEL request a SIP 2xx response is received to the SIP INVITE request, the gateway will need to send a SIP BYE request.

4)If the gateway has sent a SIP INVITE request but received no SIP response, the gateway SHALL NOT send a SIP message. If a SIP final or provisional response is subsequently received, the gateway SHALL then act in accordance with 1, 2 or 3 above respectively.

5)If the gateway has received a SIP INVITE request but not sent a SIP final response, the gateway SHALL send a SIP final response chosen according to the cause value in the received QSIG message as specified in table 1.

In all cases the gateway SHALL also disconnect media streams, if established, and allow QSIG and SIP signalling to complete in accordance with ECMA-143 and RFC-3261 respectively.

Table 1 – Mapping of QSIG Cause Value to SIP 4xx-6xx responses

QSIG Cause value	SIP response
1 Unallocated number	410 Gone
2 No route to specified transit network	404 Not found
3 No route to destination	404 Not found
4 Send special information tone	502 Bad Gateway or NA
5 Misdialled trunk prefix	410 Gone
6 Channel unacceptable	502 Bad gateway
7 Call awarded and being delivered in an established channel	502 Bad gateway
8 Preemption	502 Bad gateway
9 Preemption- circuit reserved for reuse	502 Bad gateway
16 Normal call clearing	502 Bad gateway or BYE
17 User busy	486 Busy here
18 No user responding	480 Temporarily unavailable
19 No answer from the user	480 Temporarily unavailable
20 Subscriber absent	480 Temporarily unavailable
21 Call rejected	603 Decline
22 Number changed	410 Gone
23 Redirection to new destination	410 Gone
25 Exchange routing error	502 Bad gateway
26 Non selected user clearing	502 Bad gateway or NA
27 Destination out of order	480 Temporarily unavailable
28 Address incomplete	484 Address incomplete
29 Facility rejected	488 Not Acceptable Here
30 Response to STATUS ENQUIRY	502 Bad gateway (if received in call clearing message)
31 Normal unspecified	502 Bad gateway
34 No circuit/channel available	503 Service unavailable
38 Network out of order	502 Bad gateway
39 Permanent frame mode connection out of service	502 Bad gateway or NA
40 Permanent frame mode connection operational	502 Bad gateway or NA

41 Temporary failure	503 Service unavailable
42 Switching equipment congestion	502 Bad gateway
43 Access information discarded	502 Bad gateway or NA
44 Requested circuit/channel not available	503 Service unavailable
46 Precedence call blocked	502 Bad gateway
47 Resource unavailable, unspecified	502 Bad gateway
49 Quality of service not available	503 Service unavailable
50 Requested facility not subscribed	503 Service unavailable
53 Outgoing calls barred within CUG	488 Not Acceptable Here
55 Incoming calls barred within CUG	488 Not Acceptable Here
57 Bearer capability not authorized	488 Not Acceptable Here
58 Bearer capability not presently available	503 Service unavailable
62 Inconsistency in designated outgoing access information and subscriber class	502 Bad gateway
63 Service or option not available, unspecified	503 Service unavailable
65 Bearer capability not implemented	501 Not implemented
66 Channel type not implemented	502 Bad gateway
69 Requested facility not implemented	503 Service unavailable
70 Only restricted digital information bearer capability is available	503 Service unavailable
79 Service or option not implemented, unspecified	503 Service unavailable
81 Invalid call reference value	502 Bad gateway
82 Identified channel does not exist	502 Bad gateway
83 A suspended call exists, but this call identity does not	502 Bad gateway
84 Call identity in use	502 Bad gateway
85 No call suspended	502 Bad gateway
86 Call having the requested	502 Bad gateway

call identity has been cleared	
87 User not member of CUG	488 Not Acceptable Here
88 Incompatible destination	488 Not Acceptable Here
90 Non-existant CUG	502 Bad gateway
91 Invalid transit network selection	502 Bad gateway
95 Invalid message, unspecified	500 Server internal error
96 Mandatory information element is missing	500 Server internal error
97 Message type non-existent or not implemented	500 Server internal error
98 Message not compatible with call state or message non-existent or not implemented	500 Server internal error
99 Information element non-existent or not implemented	500 Server internal error
98 Invalid information element contents	500 Server internal error
101 Message not compatible with call state	500 Server internal error
102 Recovery on timer expiry	500 Server internal error
103 Parameter non-existent or not implemented, passed on	500 Server internal error
110 Message with unrecognized parameter, discarded	500 Server internal error
111 Protocol error	500 Server internal error
127 Interworking, unspecified	500 Server internal error

8.4.2 Receipt of a SIP BYE request

On receipt of a SIP BYE request, the gateway SHALL send a QSIG DISCONNECT message with cause value 16 (normal call clearing). The gateway SHALL also disconnect media streams, if established, and allow QSIG and SIP signalling to complete in accordance with ECMA-143 and RFC-3261 respectively.

NOTE. When responding to a SIP BYE request, in accordance with RFC 3261 the gateway is also required to respond to any other outstanding transactions, e.g., with a SIP 487 (Request Terminated) response. This applies in particular if the gateway has not yet returned a final response to the SIP INVITE request.

8.4.3 Receipt of a SIP CANCEL request

On receipt of a SIP CANCEL request to clear a call for which the gateway has not sent a SIP final response to the received SIP INVITE request, the gateway SHALL send a QSIG DISCONNECT message with cause value 16 (normal call clearing). The gateway SHALL also disconnect media streams, if established, and allow QSIG and SIP signalling to complete in accordance with ECMA-143 and RFC-3261 respectively.

8.4.4 Receipt of a SIP 4xx - 6xx response

On receipt of a SIP final response (4xx-6xx) to a SIP INVITE request, the gateway SHALL transmit a QSIG DISCONNECT message. The cause value in the QSIG DISCONNECT message SHALL be derived from the SIP 4xx-6xx response according to table 2. The gateway SHALL also disconnect media streams, if established, and allow QSIG and SIP signalling to complete in accordance with ECMA-143 and RFC-3261 respectively.

Table 2 – Mapping of SIP 4xx-6xx responses to QSIG Cause values

SIP response	QSIG Cause value
400 Bad request	41 Temporary failure
401 Unauthorized	88 Incompatible destination
402 Payment required	88 Incompatible destination
403 Forbidden	88 Incompatible destination
404 Not found	3 No route to destination
405 Method not allowed	41 Temporary Failure
406 Not acceptable	41 Temporary Failure
407 Proxy Authentication required	41 Temporary Failure
408 Request timeout	41 Temporary Failure
409 Conflict	41 Temporary Failure
410 Gone	1 Unallocated number
411 Length required	41 Temporary Failure
413 Request Entity too long	41 Temporary Failure
414 Request-URI too long	41 Temporary Failure
415 Unsupported media type	65 Bearer Capability Not implemented
420 Bad extension	41 Temporary Failure
480 Temporarily unavailable	18 No user responding
481 Call/transaction doesn't exist	41 Temporary Failure
482 Loop detected	41 Temporary Failure
483 Too many hops	41 Temporary Failure
484 Address incomplete	28 Invalid number format
485 Ambiguous	1 Unallocated Number
486 Busy here	17 User busy
487 Requested Terminated	41 Temporary Failure
488 Not Acceptable Here	65 Bearer Capability Not implemented
500 Server internal error	41 Temporary Failure
501 Not implemented	41 Temporary Failure
502 Bad gateway	41 Temporary Failure

503 Service unavailable	41 Temporary Failure
504 Gateway time-out	41 Temporary Failure
505 Version not supported	41 Temporary Failure
600 Busy everywhere	17 User busy
603 Decline	21 Call rejected
604 Does not exist anywhere	1 Unallocated number
606 Not acceptable	21 Call Rejected

8.4.5 Timer expiry

The gateway SHALL run protocol timers as specified for QSIG in ECMA-143 and for SIP in RFC 3261. On expiry of these timers the actions SHALL be as specified by the respective protocol specifications.

If the call is to be cleared due to expiry of a QSIG timer, clearing of the SIP call SHALL be in accordance with 8.4.1, except that if a final response to a SIP INVITE request needs to be sent, a SIP 408 (Request Timeout) response SHALL be used. If the call is to be cleared due to expiry of a SIP timer, the gateway SHALL send a QSIG DISCONNECT message with Cause Value 41 (Temporary Failure) to clear the call in the PISN.

8.5 Request to change media characteristics

If after a call has been successfully established the gateway receives a SIP INVITE request to change the media characteristics of the call, the gateway SHALL send back a SIP 503 (Service unavailable) response and SHALL NOT change the media characteristics of the existing call.

9 Number mapping

The SIP ‘To’, ‘Request-URI’ and ‘From’ are in the form of Universal Resource Identifiers (URIs).

9.1 Mapping from SIP to QSIG

The gateway SHALL map the SIP Request-URI and From header to the QSIG Called and Calling party number information elements respectively. The way in which this is achieved is outside the scope of this specification.

The gateway SHALL set the Numbering Plan Identification (NPI) and Type of Number (TON) fields in the QSIG Called and Calling party number information elements in accordance with ECMA-155.

In the QSIG Calling party number information element, unless the gateway performs screening, the screening indicator SHALL be set to "user provided, not screened" (0). Unless the gateway performs presentation restriction, the presentation indicator SHALL be set to "presentation allowed" (0). Support of screening and/or presentation restriction is outside the scope of this specification.

Unless the gateway has a means of determining the identity of the user that answers a call from QSIG to SIP, the QSIG Connected number information SHALL NOT be generated.

9.2 Mapping from QSIG to SIP

The gateway SHALL map the QSIG Called party number information element to the SIP Request-URI and the SIP To header, both of which SHOULD contain the same value. The gateway SHALL map the QSIG Calling party number information element to the SIP From header. The way in which this is achieved is outside the scope of this specification.

If the Calling party number information element is not received in the QSIG SETUP message or if it does not contain a number or if the presentation indicator has the value "Presentation restricted", the gateway SHALL use its own address to generate the From header.

10 Requirements for support of basic services

This document specifies signalling interworking for basic services that provide a bi-directional transfer capability for speech, facsimile and modem media between the two networks.

10.1 Derivation of QSIG Bearer capability information element

The gateway SHALL generate the Bearer Capability Information Element in the QSIG SETUP message based on the SDP information received along with the SIP INVITE request. If the SIP INVITE request does not contain SDP information or the media type in the SDP is only ‘audio’ then the Bearer capability information element SHALL BE generated according to table 3. Coding of the Bearer capability information element for other media types is outside the scope of this specification.

Table 3 – Bearer capability encoding for ‘audio’ transfer

Field	Value
Coding Standard	"CCITT standardized coding" (00)
Information transfer capability	"3,1 kHz audio" (10000)
Transfer mode	"circuit mode" (00)
Information transfer rate	"64 Kbits/s" (10000)
Multiplier	Octet omitted
User information layer 1	Generated by gateway based on

protocol	information of the PISN. Supported values are "CCITT recommendation G.711 µ-law" (00010) "CCITT recommendation G.711 A-law" (00011)

10.2 Derivation of media type in SDP

The gateway SHALL generate the SDP information to include in the SIP INVITE request based on the Bearer capability information element received in the QSIG SETUP message. The media type included in the SDP SHALL be according to table 4.

Table 4 – Media type setting in SDP based on Bearer capability information element

Information transfer capability in Bearer capability information element	Media type in SDP
"speech" (00000)	audio
"3,1 kHz audio" (10000)	audio
"unrestricted digital information" (01000)	data

11 Security considerations

The security considerations of RFC 3261 apply.

12 Author's Addresses

John Elwell
Siemens Communications Limited
Technology Drive
Beeston
Nottingham, UK, NG9 1LA
email: john.elwell@siemenscomms.co.uk

Frank Derks
Philips Business Communications
P.O. Box 32
1200 JD, Hilversum
The Netherlands
email: frank.derks@philips.com

Olivier Rousseau
Alcatel Business Systems
32,Avenue Kleber

92700 Colombes
France
email: olivier.rousseau@col.bsf.alcatel.fr

Patrick Mourot
Alcatel Business Systems
1,Rue Dr A. Schweitzer
67400 Illkirch
France
email: patrick.mourot@sxb.bsf.alcatel.fr

13 Normative References

[1] ECMA-133 "Private Integrated Services Network (PISN – Reference configuration for PISN exchanges (PINX)" (International Standard ISO/IEC 11579-1)

[2] ECMA-143 "Private Integrated Services Network - Circuit-mode Bearer Services - Inter-Exchange Signalling Procedures and Protocol" (International Standard ISO/IEC 11572)

[3] ECMA-165 "Private Integrated Services Network - Generic Functional Protocol for the Support of Supplementary Services - Inter-Exchange Signalling Procedures and Protocol" (International Standard ISO/IEC 11582)

[4] ECMA-307 "Corporate Telecommunication Networks - Signalling Interworking between QSIG and H.323 - Generic Functional Protocol for the Support of Supplementary Services" (International Standard ISO/IEC 21409)

[5] J. Postel, " Transmission Control Protocol", RFC 793.

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

[7] T. Dierks, C.Allen, "The TLS protocol version 1.0", RFC 2246.

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

[9] R. Stewart et al., "Stream Control Transmission Protocol" RFC 2960.

[10] J. Rosenberg, H. Schulzrinne, et al., "SIP: Session initiation protocol", RFC 3261, April 2002.

[11] J. Rosenberg, H. Schulzrinne, "Reliability of Provisional Responses in SIP", RFC BBBB.

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Annex A (informative) – Example message sequences

A.1 Introduction

This annex shows some typical message sequences that can occur for an interworking between QSIG and SIP.

NOTE. For all message sequence diagrams, there is no message mapping between QSIG and SIP unless explicitly indicated by dotted lines. Also, if there are no dotted lines connecting two messages, this means that these are independent of each other in terms of the time when they occur.

NOTE. Numbers prefixing SIP method names and response codes in the diagrams represent sequence numbers. Messages bearing the same number will have the same value in the CSeq header.

A.2 Message sequences for call establishment from QSIG to SIP

Below are typical message sequences for successful call establishment from QSIG to SIP

```
                           .-------------------.
                           |                   |
                           |      GATEWAY      |
 PISN                      |                   |         IP NETWORK
   |                       `-----+------+------'                  |
   |                             |      |                         |
   |                             |      |                         |
   |   QSIG SETUP                |      |        1-INVITE         |
  1|---------------------------->|......|------------------------>| 2
   |                             |      |                         |
   |                             |      |                         |
   | QSIG CALL PROCCEEDING       |      |        1-100 TRYING     |
  3|<----------------------------|      |<------------------------+ 4
   |                             |      |                         |
   |                             |      |                         |
   |   QSIG ALERTING             |      |        1-180 RINGING    |
  8|<----------------------------|......|<------------------------+ 5
   |                             |      |                         |
   |                             |      |        2-PRACK          |
   |                             |      |------------------------>| 6
   |                             |      |        2-200 OK         |
   |                             |      |<------------------------+ 7
   |                             |      |                         |
   |   QSIG CONNECT              |      |        1-200 OK         |
 11|<----------------------------|......|<------------------------+ 9
   |                             |      |                         |
   |   QSIG CONNECT ACK          |      |        1-ACK            |
 12|---------------------------->|      |------------------------>| 10
   |                             |      |                         |
   |<===========================>|      |<=======================>|
   |         AUDIO               |      |          AUDIO          |
```

Figure 3 – Typical message sequence for successful call establishment from QSIG to SIP using enbloc procedures on both QSIG and SIP

1 The PISN sends a QSIG SETUP message to the gateway to begin a session with a SIP UA
2 On receipt of the QSIG SETUP message, the gateway generates a SIP INVITE request and sends it to an appropriate SIP entity in the IP network based on the called number
3 The gateway sends a QSIG CALL PROCEEDING message to the PISN - no more QSIG INFORMATION messages will be accepted
4 The IP network sends a SIP 100 (Trying) response to the gateway
5 The IP network sends a SIP 180 (Ringing) response.
6 The gateway may send back a SIP PRACK request to the IP network based on the inclusion of a Require header or a Supported header with option tag 100rel in the initial SIP INVITE request
7 The IP network sends a SIP 200 (OK) response to the gateway to acknowledge the SIP PRACK request
8 The gateway maps this SIP 180 (Ringing) response to a QSIG ALERTING message and sends it to the PISN.
9 The IP network sends a SIP 200 (OK) response when the call is

answered.
10 The gateway sends a SIP ACK request to acknowledge the SIP 200 (OK)response.
11 The gateway maps this SIP 200 (OK) response to a QSIG CONNECT message and sends it to the PISN.
12 The PISN sends a QSIG CONNECT ACKNOWLEDGE message in response to the QSIG CONNECT message.

```
                      +------------------------+
   PISN               |        GATEWAY         |         IP NETWORK
                      |                        |
   |  QSIG SETUP      +--------+-------+-------+                 |
  1|-------------------------->|       |                         |
   |                           |       |                         |
   |  QSIG SETUP ACK           |       |                         |
  2|<--------------------------|       |                         |
   |                           |       |                         |
   | QSIG INFORMATION          |       |                         |
  3|-------------------------->|       |                         |
   |                           |       |                         |
   | QSIG INFORMATION          |       |  1-INVITE               |
 3a|-------------------------->|.......|------------------------>|4
   | QSIG CALL PROCEEDING      |       |  1-100 TRYING           |
  5|<--------------------------|       |<------------------------|6
   |                           |       |                         |
   | QSIG ALERTING             |       |  1-180 RINGING          |
 10|<--------------------------|.......|<------------------------|7
   |                           |       |  2-PRACK                |
   |                           |       |------------------------>|8
   |                           |       |  2-200 OK               |
   |                           |       |<------------------------|9
   | QSIG CONNECT              |       |  1-200 OK               |
 13|<--------------------------|.......|<------------------------|11
   |                           |       |                         |
   | QSIG CONNECT ACK          |       |  1-ACK                  |
 14|-------------------------->|       |------------------------>|12
   |          AUDIO            |       |           AUDIO         |
   |<=========================>|       |<=======================>|
```

Figure 4 – Typical message sequence for successful call establishment from QSIG to SIP using overlap receiving on QSIG and enbloc sending on SIP

1 The PISN sends a QSIG SETUP message to the gateway to begin a session with a SIP UA. The QSIG SETUP message does not contain a Sending Complete information element.
2 The gateway sends a QSIG SETUP ACKNOWLEDGE message to the PISN. More digits are expected.
3 More digits are sent from the PISN within a QSIG INFORMATION message.

3a More digits are sent from the PISN within a QSIG INFORMATION message. The QSIG INFORMATION message contains a Sending Complete information element
4 The Gateway generates a SIP INVITE request and sends it to an appropriate SIP entity in the IP network, based on the called number
5 The gateway sends a QSIG CALL PROCEEDING message to the PISN - no more QSIG INFORMATION messages will be accepted
6 The IP network sends a SIP 100 (Trying) response to the gateway
7 The IP network sends a SIP 180 (Ringing) response.
8 The gateway may send back a SIP PRACK request to the IP network based on the inclusion of a Require header or a Supported header with option tag 100rel in the initial SIP INVITE request
9 The IP network sends a SIP 200 (OK) response to the gateway to acknowledge the SIP PRACK request
10 The gateway maps this SIP 180 (Ringing) response to a QSIG ALERTING message and sends it to the PINX.
11 The IP network sends a SIP 200 (OK) response when the call is answered.
12 The gateway sends an SIP ACK request to acknowledge the SIP 200 (OK) response.
13 The gateway maps this SIP 200 (OK) response to a QSIG CONNECT message and sends it to the PINX.
14 The PISN sends a QSIG CONNECT ACKNOWLEDGE message in response to the QSIG CONNECT message.

```
                      +----------------------+
  PISN                |        GATEWAY       |           IP NETORK
                      |                      |
  |  QSIG SETUP       +-------+-------+------+                   |
1 |-------------------------->|       |                          |
  |                           |       |                          |
  |  QSIG SETUP ACK           |       |                          |
2 |<--------------------------|       |                          |
  |                           |       |                          |
  | QSIG INFORMATION          |       |                          |
3 |-------------------------->|       |                          |
  | QSIG INFORMATION          |       | 1-INVITE                 |
3 |-------------------------->|.......|------------------------->|4
  |                           |       | 1-484                    |
  |                           |       |<-------------------------|5
  |                           |       | 1-ACK                    |
  |                           |       |------------------------->|6
  | QSIG INFORMATION          |       | 2-INVITE                 |
7 |-------------------------->|.......|------------------------->|4
  |                           |       | 2-484                    |
  |                           |       |<-------------------------|5
  |                           |       | 2-ACK                    |
  |                           |       |------------------------->|6
  |                           |       |                          |
  | QSIG INFORMATION          |       |                          |
  | Sending Complete IE       |       | 3-INVITE                 |
8 |-------------------------->|.......|------------------------->|10
  | QSIG CALL PROCEEDING      |       | 3-100 TRYING             |
9 |<--------------------------|       |<-------------------------|11
  |                           |       |                          |
  | QSIG ALERTING             |       | 3-180 RINGING            |
15|<--------------------------|.......|<-------------------------|12
  |                           |       | 4-PRACK                  |
  |                           |       |------------------------->|13
  |                           |       | 4-200 OK                 |
  |                           |       |<-------------------------|14
  | QSIG CONNECT              |       | 3-200 OK                 |
18|<--------------------------|.......|<-------------------------|16
  |                           |       |                          |
  | QSIG CONNECT ACK          |       | 3-ACK                    |
19|-------------------------->|       |------------------------->|17
  |         AUIO              |       |         AUDIO            |
  |<=========================>|       |<========================>|
  |                           |       |                          |
```

Figure 5 – Typical message sequence for successful call establishment from QSIG to SIP using overlap procedures on both QSIG and SIP

1 The PISN sends a QSIG SETUP message to the gateway to begin a session with a SIP UA. The QSIG SETUP message does not contain a Sending complete information element.
2 The gateway sends a QSIG SETUP ACKNOWLEDGE message to the PISN. More digits are expected.

3 More digits are sent from the PISN within a QSIG INFORMATION message.
4 When the gateway receives the minimum number of digits required to route the call it generates a SIP INVITE request and sends it to an appropriate SIP entity in the IP network based on the called number
5 Due to an insufficient number of digits the IP network will return a SIP 484 (Address Incomplete) response.
6 The SIP 484 (Address Incomplete) response is acknowledged.
7 More digits are received from the PISN in a QSIG INFORMATION message. A new INVITE is sent with the same Call-ID but an updated Request-URI.
8 More digits are received from the PISN in a QSIG INFORMATION message. The QSIG INFORMATION message contains a Sending Complete information element
9 The gateway sends a QSIG CALL PROCEEDING message to the PISN - no more information will be accepted
10 The gateway sends a new SIP INVITE request with an updated Request-URI field.
11 The IP network sends a SIP 100 (Trying) response to the gateway
12 The IP network sends a SIP 180 (Ringing) response.
13 The gateway may send back a SIP PRACK request to the IP network based on the inclusion of a Require header or a Supported header with option tag 100rel in the initial SIP INVITE request
14 The IP network sends a SIP 200 (OK) response to the gateway to acknowledge the SIP PRACK request
15 The gateway maps this SIP 180 (Ringing) response to a QSIG ALERTING message and sends it to the PISN.
16 The IP network sends a SIP 200 (OK) response when the call is answered.
17 The gateway sends a SIP ACK request to acknowledge the SIP 200 (OK) response.
18 The gateway maps this SIP 200 (OK) response to a QSIG CONNECT message.
19 The PISN sends a QSIG CONNECT ACKNOWLEDGE message in response to the QSIG CONNECT message.

A.3 Message sequences for call establishment from SIP to QSIG

Below are typical message sequences for successful call establishment from SIP to QSIG

```
                         +----------------------+
     IP NETWORK          |        GATEWAY       |                PISN
                         |                      |
     |                   +-------+-------+------+                   |
     |                           |       |                          |
     |                           |       |                          |
     |     1-INVITE              |       | QSIG SETUP               |
   1 |-------------------------->|.......|------------------------->|3
     |     1-100 TRYING          |       | QSIG CALL PROCEEDING     |
   2 |<--------------------------|       |<-------------------------|4
     |     1-180 RINGING         |       | QSIG ALERTING            |
   6 |<--------------------------|.......|<-------------------------|5
     |                           |       |                          |
     |                           |       |                          |
     |     2-PRACK               |       |                          |
   7 |-------------------------->|       |                          |
     |     2-200 OK              |       |                          |
   8 |<--------------------------|       |                          |
     |     1-200 OK              |       | QSIG CONNECT             |
   11|<--------------------------|.......|<-------------------------|9
     |                           |       |                          |
     |     1-ACK                 |       | QSIG CONNECT ACK         |
   12|-------------------------->|       |------------------------->|10
     |         AUDIO             |       |         AUDIO            |
     |<=========================>|       |<========================>|
     |                           |       |                          |
```

Figure 6 – Typical message sequence for successful call establishment from SIP to QSIG using enbloc procedures

1 The IP network sends a SIP INVITE request to the gateway
2 The gateway sends a SIP 100 (Trying) response to the IP network
3 On receipt of the SIP INVITE request, the gateway sends a QSIG SETUP message
4 The PISN sends a QSIG CALL PROCEEDING message to the gateway
5 A QSIG ALERTING message is returned to indicate that the end user in the PISN is being alerted
6 The gateway maps the QSIG ALERTING message to a SIP 180 (Ringing) response
7 The IP network can send back a SIP PRACK request to the IP network based on the inclusion of a Require header or a Supported header with option tag 100rel in the initial SIP INVITE request
8 The gateway sends a SIP 200 (OK) response to acknowledge the SIP PRACK request
9 The PISN sends a QSIG CONNECT message to the gateway when the call is answered
10 The gateway sends a QSIG CONNECT ACKNOWLEDGE message to acknowledge the QSIG CONNECT message
11 The QSIG CONNECT message is mapped to a SIP 200 (OK) response.
12 The IP network, upon receiving a SIP INVITE final response (200), will send a SIP ACK request to acknowledge receipt

```
                      +-----------------------+
  IP NETWORK          |        GATEWAY        |               PISN
                      |                       |
  | 1-INVITE          +-------+-------+-------+                  |
1 |-------------------------->|       |                          |
  |     1-484                 |       |                          |
2 |<--------------------------|       |                          |
  |     1-ACK                 |       |                          |
3 |-------------------------->|       |                          |
  |     2-INVITE              |       |                          |
1 |-------------------------->|       |                          |
  |     2-484                 |       |                          |
2 |<--------------------------|       |                          |
  |     2- ACK                |       |                          |
3 |-------------------------->|       |                          |
  |     3-INVITE              |       | QSIG SETUP               |
4 |-------------------------->|.......|------------------------->|6
  |     3-100 TRYING          |       | QSIG CALL PROCEEDING     |
5 |<--------------------------|       |<-------------------------|7
  |     3-180 RINGING         |       | QSIG ALERTING            |
9 |<--------------------------|.......|<-------------------------|8
  |                           |       |                          |
  |                           |       |                          |
  |     4-PRACK               |       |                          |
10|-------------------------->|       |                          |
  |     5-200 OK              |       |                          |
11|<--------------------------|       |                          |
  |     3-200 OK              |       | QSIG CONNECT             |
14|<--------------------------|.......|<-------------------------|12
  |                           |       |                          |
  |     3-ACK                 |       | QSIG CONNECT ACK         |
15|-------------------------->|       |------------------------->|13
  |         AUDIO             |       |         AUDIO            |
  |<=========================>|       |<========================>|
  |                           |       |                          |
```

Figure 7 – Typical message sequence for successful call establishment from SIP to QSIG using overlap receiving on SIP and enbloc sending on QSIG

1 The IP network sends a SIP INVITE request to the gateway
2 Due to an insufficient number of digits the gateway returns a SIP 484(Address Incomplete) response.
3 The IP network acknowledge the SIP 484 (Address Incomplete) response.
4 The IP network sends a new SIP INVITE request with the same Call-ID and updated Request-URI.
5 The gateway now has all the digits required to route the call to the PISN. The gateway sends back a SIP 100 (Trying) response
6 The gateway sends a QSIG SETUP message
7 The PISN sends a QSIG CALL PROCEEDING message to the gateway

8 A QSIG ALERTING message is returned to indicate that the end user in the PISN is being alerted
9 The gateway maps the QSIG ALERTING message to a SIP 180 (Ringing)response
10 The IP network can send back a SIP PRACK request to the IP network based on the inclusion of a Require header or a Supported header with option tag 100rel in the initial SIP INVITE request
11 The gateway sends a SIP 200 (OK) response to acknowledge the SIP PRACK request
12 The PISN sends a QSIG CONNECT message to the gateway when the call is answered
13 The gateway sends a QSIG CONNECT ACKNOWLEDGE message to acknowledge the CONNECT message
14 The QSIG CONNECT message is mapped to a SIP 200 (OK) response.
15 The IP network, upon receiving a SIP INVITE final response (200), will send a SIP ACK request to acknowledge receipt

```
                      +----------------------+
  IP NETWORK          |        GATEWAY       |                PISN
                      |                      |
  | 1-INVITE          +-------+-------+------+                   |
1 |-------------------------->|       |                          |
  |     1-484                 |       |                          |
2 |<--------------------------|       |                          |
  |     1-ACK                 |       |                          |
3 |-------------------------->|       |                          |
  |     2-INVITE              |       | QSIG SETUP               |
4 |-------------------------->|.......|------------------------->|6
  |     2-100 TRYING          |       | QSIG SETUP ACK           |
5 |<--------------------------|       |<-------------------------|7
  |     3- INVITE             |       | QSIG INFORMATION         |
8 |-------------------------->|.......|------------------------->|10
  |     3-100 TRYING          |       |                          |
9 |<--------------------------|       | QSIG CALL PROCEEDING     |
  |                           |       |<-------------------------|11
13|     3-180 RINGING         |       | QSIG ALERTING            |
  |<--------------------------|.......|<-------------------------|12
  |     2-484                 |       |                          |
14|<--------------------------|       |                          |
  |     2-ACK                 |       |                          |
15|-------------------------->|       |                          |
  |     4-PRACK               |       |                          |
16|-------------------------->|       |                          |
  |     4-200 OK              |       |                          |
17|<--------------------------|       |                          |
  |     3-200 OK              |       | QSIG CONNECT             |
20|<--------------------------|.......|<-------------------------|18
  |                           |       |                          |
  |     3-ACK                 |       | QSIG CONNECT ACK         |
21|-------------------------->|       |------------------------->|19
  |         AUDIO             |       |         AUDIO            |
  |<=========================>|       |<========================>|
  |                           |       |                          |
```

Figure 8 – Typical message sequence for successful call establishment from SIP to QSIG using overlap procedures on both SIP and QSIG

1 The IP network sends a SIP INVITE request to the gateway
2 Due to an insufficient number of digits the gateway returns a SIP 484(Address Incomplete) response.
3 The IP network acknowledge the SIP 484 (Address Incomplete) response.
4 The IP network sends a new SIP INVITE request with the same Call-ID and updated Request-URI.
5 The gateway now has all the digits required to route the call to the PISN. The gateway sends back a SIP 100 (Trying) response to the IP network
6 The gateway sends a QSIG SETUP message

7 The PISN needs more digits to route the call and sends a QSIG SETUP ACKNOWLEDGE message to the gateway
8 The IP network sends a new SIP INVITE request with the same Call-ID and updated Request-URI.
9 The gateway sends back a SIP 100 (Trying) response to the IP network
10 The gateway maps the new SIP INVITE request to a QSIG INFORMATION message
11 The PISN has all the digits required and sends back a QSIG CALL PROCEEDING message to the gateway
12 A QSIG ALERTING message is returned to indicate that the end user in the PISN is being alerted
13 The gateway maps the QSIG ALERTING message to a SIP 180 (Ringing)response
14 The gateway sends a SIP 484 (Address Incomplete) response for the previous SIP INVITE request
15 The IP network acknowledges the SIP 484 (Address Incomplete) response
16 The IP network can send back a SIP PRACK request to the IP network based on the inclusion of a Require header or a Supported header with option tag 100rel in the initial SIP INVITE request
17 The gateway sends a SIP 200 (OK) response to acknowledge the SIP PRACK request
18 The PISN sends a QSIG CONNECT message to the gateway when the call is answered
19 The gateway sends a QSIG CONNECT ACKNOWLEDGE message to acknowledge the QSIG CONNECT message
20 The QSIG CONNECT message is mapped to a SIP 200 (OK) response.
21 The IP network, upon receiving a SIP INVITE final response (200), will send a SIP ACK request to acknowledge receipt

A.4 Message sequence for call clearing from QSIG to SIP

Below are typical message sequences for Call Clearing from QSIG to SIP

```
                         .-------------------.
                         |                   |
                         |      GATEWAY      |
     PISN                |                   |          IP NETWORK
       |                 `-----+------+------'                |
       |                       |      |                       |
       |                       |      |                       |
       |     QSIG DISCONNECT   |      |   2- BYE              |
      1|---------------------->|......|---------------------->|4
       |     QSIG RELEASE      |      |        2-200 OK       |
      2|<----------------------|      |<----------------------|5
       |     QSIG RELEASE COMP |      |                       |
      3|---------------------->|      |                       |
       |                       |      |                       |
       |                       |      |                       |
       |                       |      |                       |
```

Figure 9 – Typical message sequence for call clearing from QSIG to SIP subsequent to call establishment

1 The PISN sends a QSIG DISCONNECT message to the gateway
2 The gateway sends back a QSIG RELEASE message to the PISN in response to the QSIG DISCONNECT message
3 The PISN sends a QSIG RELEASE COMPLETE message in response. All PISN resources are now released.
4 The gateway maps the QSIG DISCONNECT message to a SIP BYE request
5 The IP network sends back a SIP 200 (OK) response to the SIP BYE request. All IP resources are now released

```
                         .-------------------.
                         |                   |
                         |      GATEWAY      |
     PISN                |                   |          IP NETWORK
       |                 `-----+------+------'                |
       |                       |      |                       |
       |                       |      |                       |
       |     QSIG DISCONNECT   |      |   1- 4XX / 6XX        |
      1|---------------------->|......|---------------------->|4
       |     QSIG RELEASE      |      |        1- ACK         |
      2|<----------------------|      |<----------------------|5
       |     QSIG RELEASE COMP |      |                       |
      3|---------------------->|      |                       |
       |                       |      |                       |
       |                       |      |                       |
```

Figure 10 – Typical message sequence for call clearing from QSIG to SIP during establishment of a call from SIP to QSIG (gateway has not sent a final response to the SIP INVITE request)

1 The PISN sends a QSIG DISCONNECT message to the gateway
2 The gateway sends back a QSIG RELEASE message to the PISN in response to the QSIG DISCONNECT message

3 The PISN sends a QSIG RELEASE COMPLETE message in response. All PISN resources are now released.
4 The gateway maps the QSIG DISCONNECT message to a SIP 4xx-6xx response
5 The IP network sends back a SIP ACK request in response to the SIP 4xx-6xx response. All IP resources are now released

```
                      .-------------------.
                      |                   |
                      |      GATEWAY      |
   PISN               |                   |        IP NETWORK
     |                `-----+------+------'                 |
     |                      |      |                        |
     |                      |      |                        |
     |     QSIG DISCONNECT  |      |   1- CANCEL            |
    1|--------------------->|......|----------------------->|4
     |     QSIG RELEASE     |      |1-487 Request Terminated|
    2|<---------------------|      |<-----------------------|5
     |     QSIG RELEASE COMP|      |                        |
    3|--------------------->|      |   1- 200 OK            |
     |                      |      |<-----------------------|6
     |                      |      |                        |
     |                      |      |                        |
```

Figure 11 – Typical message sequence for call clearing from QSIG to SIP during establishment of a call from QSIG to SIP (gateway has received a provisional response to the SIP INVITE request but not a final response)

1 The PISN sends a QSIG DISCONNECT message to the gateway
2 The gateway sends back a QSIG RELEASE message to the PISN in response to the QSIG DISCONNECT message
3 The PISN sends a QSIG RELEASE COMPLETE message in response. All PISN resources are now released.
4 The gateway maps the QSIG DISCONNECT message to a SIP CANCEL request(subject to a provisional response but no final response having been received)
5 The IP network sends back a SIP 487 (Request Terminated) response to the SIP INVITE request.
6 The IP network sends back a SIP 200 (OK) response to the SIP CANCEL request. All IP resources are now released

A.5 Message sequence for call clearing from SIP to QSIG

Below are typical message sequences for Call Clearing from SIP to QSIG

```
                         .-----------------.
                         |                 |
                         |     GATEWAY     |
        IP NETWORK       |                 |          PISN
         |               `----+-----+------'            |
         |                    |     |                   |
         |                    |     |                   |
         |  2- BYE            |     |  QSIG DISCONNECT  |
        1|------------------->|.....|------------------>|3
         |                    |     |  QSIG RELEASE     |
         |                    |     |<------------------|4
         |      2-200 OK      |     |  QSIG RELEASE COMP|
        2|<-------------------|     |------------------>|5
         |                    |     |                   |
         |                    |     |                   |
```

Figure 12 – Typical message sequence for call clearing from SIP to QSIG subsequent to call establishment

1 The IP network sends a SIP BYE request to the gateway
2 The gateway sends back a SIP 200 (OK) response to the SIP BYE request. All IP resources are now released
3 The gateway maps the SIP BYE request to a QSIG DISCONNECT message
4 The PISN sends back a QSIG RELEASE message to the gateway in response to the QSIG DISCONNECT message
5 The gateway sends a QSIG RELEASE COMPLETE message in response. All PISN resources are now released.

```
                         .-----------------.
                         |                 |
                         |     GATEWAY     |
        IP NETWORK       |                 |          PISN
         |               `----+-----+------'            |
         |                    |     |                   |
         |                    |     |                   |
         |  1- 4XX / 6XX      |     |  QSIG DISCONNECT  |
        1|------------------->|.....|------------------>|3
         |                    |     |  QSIG RELEASE     |
         |                    |     |<------------------|4
         |      1- ACK        |     |  QSIG RELEASE COMP|
        2|<-------------------|     |------------------>|5
         |                    |     |                   |
         |                    |     |                   |
         |                    |     |                   |
```

Figure 13 – Typical message sequence for call clearing from SIP to QSIG during establishment of a call from QSIG to SIP (gateway has not previously received a final response to the SIP INVITE request)

1 The IP network sends a SIP 4xx-6xx response to the gateway
2 The gateway sends back a SIPACK request in response to the SIP 4xx-6xx response. All IP resources are now released

3 The gateway maps the SIP 4xx-6xx response to a QSIG DISCONNECT message
4 The PISN sends back a QSIG RELEASE message to the gateway in response to the QSIG DISCONNECT message
5 The gateway sends a QSIG RELEASE COMPLETE message in response. All PISN resources are now released.

```
                             .----------------.
                             |                |
                             |    GATEWAY     |
     IP NETWORK              |                |           PISN
       |                     `-----+-----+----'              |
       |                           |     |                   |
       |                           |     |                   |
       |  1- CANCEL                |     |    QSIG DISCONNECT |
      1|-------------------------->|.....|------------------->|4
       |                           |     |    QSIG RELEASE    |
       |                           |     |<-------------------|5
       |1-487 Request Terminated   |     |    QSIG RELEASE COMP|
      2|<--------------------------|     |------------------->|6
       |                           |     |                   |
       |  1- 200 OK                |     |                   |
      3|<--------------------------|     |                   |
```

Figure 14 – Typical message sequence for call clearing from SIP to QSIG during establishment of a call from QSIG to SIP (gateway has received a provisional response to the SIP INVITE request but not a final response)

1 The IP network sends a SIP CANCEL request to the gateway
2 The gateway sends back a SIP 487 (Request Terminated) response to the SIP INVITE request
3 The gateway sends back a SIP 200 (OK) response to the SIP CANCEL request. All IP resources are now released
4 The gateway maps the SIP 4xx-6xx response to a QSIG DISCONNECT message
5 The PISN sends back a QSIG RELEASE message to the gateway in response to the QSIG DISCONNECT message
6 The gateway sends a QSIG RELEASE COMPLETE message in response. All PISN resources are now released.