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Terrestrial Trunked Radio (TETRA);
Voice plus Data (V+D);
Part 3: Interworking at the Inter-System Interface (ISI);
Sub-part 11: General design, SIP/IP

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Foreword

This European Standard (EN) has been produced by ETSI Technical Committee TETRA and Critical Communications Evolution (TCCE).

The present document is part 3, sub-part 11 of a multi-part deliverable covering the Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D), as identified below:

Part 1: "General network design";

Part 2: "Air Interface (AI)";

Part 3: "Interworking at the Inter-System Interface (ISI)":

Sub-part 1: "General design";

Sub-part 2: "Additional Network Feature Individual Call (ANF-ISIIC)";

Sub-part 3: "Additional Network Feature Group Call (ANF-ISIGC)";

Sub-part 4: "Additional Network Feature Short Data Service (ANF-ISISDS)";

Sub-part 5: "Additional Network Feature for Mobility Management (ANF-ISIMM)";

Sub-part 6: "Speech format implementation for circuit mode transmission";

Sub-part 7: "Speech Format Implementation for Packet Mode Transmission";

Sub-part 8: "Generic Speech Format Implementation";

Sub-part 9: "Transport layer independent, General design";

Sub-part 10: "General design, PSS1 over E.1";

Sub-part 11: "General design, SIP/IP";

Sub-part 12: "Transport layer independent Additional Network Feature Individual Call (ANF-ISIIC)";

Sub-part 13: "Transport layer independent Additional Network Feature Group Call (ANF-ISIGC)";

Sub-part 14: "Transport layer independent Additional Network Feature Short Data Service (ANF-ISISDS)";

Sub-part 15: "Transport layer independent Additional Network Feature, Mobility Management (ANF-ISIMM)";

Part 4: "Gateways basic operation";

Part 5: "Peripheral Equipment Interface (PEI)";

Part 7: "Security";

Part 9: "General requirements for supplementary services";

Part 10: "Supplementary services stage 1";

Part 11: "Supplementary services stage 2";

Part 12: "Supplementary services stage 3";

Part 13: "SDL model of the Air Interface (AI)";

Part 14: "Protocol Implementation Conformance Statement (PICS) proforma specification";

Part 15: "TETRA frequency bands, duplex spacings and channel numbering";

Part 16: "Network Performance Metrics";

Part 17: "TETRA V+D and DMO specifications";

Part 18: "Air interface optimized applications";

Part 19: "Interworking between TETRA and Broadband systems".

NOTE 1: Part 3, sub-parts 6 and 7 (Speech format implementation), part 4, sub-part 3 (Data networks gateway), part 10, sub-part 15 (Transfer of control), part 13 (SDL) and part 14 (PICS) of this multi-part deliverable are in status "historical" and are not maintained.

NOTE 2: Some parts are also published as Technical Specifications such as ETSI TS 100 392-2 and those may be the latest version of the document.

For all subparts in the TETRA specification ETSI EN 300 392-3 [i.1], "Interworking at the Inter-System Interface (ISI)" the terms ISI and TETRA ISI are equivalent.

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Modal verbs terminology

In the present document "**shall**", "**shall not**", "**should**", "**should not**", "**may**", "**need not**", "**will**", "**will not**", "**can**" and "**cannot**" are to be interpreted as described in clause 3.2 of the ETSI Drafting Rules (Verbal forms for the expression of provisions).

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1 Scope

The present document defines the transport of Inter System Interface (ISI) PDUs using the Session Initiation Protocol (SIP) as transport layer and the Session Description Protocol (SDP) as media control protocol. It specifies:

- SIP methods and header fields used for transport of ISI PDUs; and

- the SDP content; and

- use the ISI Mediation Function which coordinates the communication between TETRA systems.

The ISI Mediation Function specification applies to an IP based TETRA Switching and Management Infrastructure (SwMI) which supports the ISI.

2 References

2.1 Normative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the referenced document (including any amendments) applies.

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The following referenced documents are necessary for the application of the present document.

[1] Void.

[2] ETSI EN 300 392-3-8: "Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D); Part 3: Interworking at the Inter-System Interface (ISI); Sub-part 8: Generic Speech Format Implementation".

[3] ETSI EN 300 392-3-9: "Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D); Part 3: Interworking at the Inter-System Interface (ISI); Sub-part 9: Transport layer independent, General design".

[4] ETSI EN 300 392-3-12: "Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D); Part 3: Interworking at the Inter-System Interface (ISI); Sub-part 12: Transport Layer Independent Additional Network Feature Individual Call (ANF-ISIIC)".

[5] ETSI EN 300 392-3-13: "Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D); Part 3: Interworking at the Inter-System Interface (ISI); Sub-part 13: Transport layer independent Additional Network Feature Group Call (ANF-ISIGC)".

[6] ETSI EN 300 392-3-14: "Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D); Part 3: Interworking at the Inter-System Interface (ISI); Sub-part 14: Transport layer independent Additional Network Feature Short Data Service (ANF-ISISDS)".

[7] ETSI EN 300 392-3-15: "Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D); Part 3: Interworking at the Inter-System Interface (ISI); Sub-part 15: Transport layer independent Additional Network Feature, Mobility Management (ANF-ISIMM)".

[8] IETF RFC 3261: "SIP: Session Initiation Protocol".

[9]   IETF RFC 3264: "An Offer/Answer Model with Session Description Protocol (SDP)".

[10]   IETF RFC 3428: "Session Initiation Protocol (SIP) Extension for Instant Messaging".

[11]   IETF RFC 4028: "Session Timers in the Session Initiation Protocol (SIP)".

[12]   IETF RFC 4566: "SDP: Session Description Protocol".

[13]   IETF RFC 5621: "Message Body Handling in the Session Initiation Protocol (SIP)".

[14]   IETF RFC 6026: "Correct Transaction Handling for 2xx Responses to Session Initiation Protocol (SIP) INVITE Requests".

[15]   IETF RFC 6086: "Session Initiation Protocol (SIP) INFO Method and Package Framework".

[16]   IETF RFC 6141: "Re-INVITE and Target-Refresh Request Handling in the Session Initiation Protocol (SIP)".

[17]   IETF RFC 2045: "Multipurpose Internet Mail Extensions (MIME) Part One: Format of Internet Message Bodies".

2.2   Informative references

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The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

[i.1]   ETSI EN 300 392-3: "Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D); Part 3: Interworking at the Inter-System Interface (ISI)".

3   Definition of terms, symbols and abbreviations

3.1   Terms

For the purposes of the present document, the following terms apply:

destination SwMI: Switching and Management Infrastructure where the receiving ANF-ISI entity is located (in the context of a single one-way exchange of information between two ANF-ISI entities located in different Switching and Management Infrastructures)

group attached SwMI: Switching and Management Infrastructure different from the home SwMI of the group considered in which at least one individual subscriber member of the group is attached to that group

Group TETRA Subscriber Identity (GTSI): TETRA Subscriber Identity assigned to a group

home SwMI: Switching and Management Infrastructure in which the subscription of a given user is registered

NOTE: That user is defined as being a subscriber (see below the definition of that term).

invocation: action taken by the user or by the service provider to execute a specific service function within real time

Location Area (LA): area within radio coverage of a base station or group of base stations within which a Mobile Station (MS) is allowed to operate

Mobile Network Identity (MNI): identity that is broadcast by all TETRA base stations to uniquely identify the SwMI

NOTE: It consists of the Mobile Country Code (MCC) and the Mobile Network Code (MNC).

Mobile Station (MS): physical grouping that contains all of the mobile equipment that is used to obtain TETRA services

NOTE: By definition, a mobile station contains at least one Mobile Radio Stack (MRS).

originating SwMI: in the context of a TETRA call, Switching and Management Infrastructure where the calling user is registered (which implies that this user is located in that SwMI) or Switching and Management Infrastructure which originates a Call independent signalling connection

service user: abstract representation of the totality of those entities in a single system that makes use of a service through a single access point

Short Subscriber Identity (SSI): network specific portion of a TSI

NOTE: A SSI is only unique within one TETRA sub-domain (one TETRA network).

source SwMI: Switching and Management Infrastructure where the sending ANF-ISI entity is located (in the context of a single one-way exchange of information between two ANF-ISI entities located in different Switching and Management Infrastructures)

subscriber: user of a telecommunication service, based on a contract with the provider of the service

NOTE 1: The subscriber may be an individual or a group: in the first case it is identified by an ITSI, in the second, by a GTSI.

NOTE 2: The individual subscriber is able to access an SwMI either through a MS or Line Station.

supplementary service: service which modifies or supplements a basic bearer service or a basic teleservice

NOTE: A supplementary service cannot be offered to a customer as a stand-alone service. It should be offered in combination with a bearer service or a teleservice.

Switching and Management Infrastructure (SwMI): all of the TETRA equipment for a Voice plus Data (V+D) network

NOTE: The SwMI enables users to communicate with each other.

terminating SwMI: in the context of a TETRA call, Switching and Management Infrastructure where the calling user is registered (which implies that this user is located in that SwMI) or Switching and Management Infrastructure which terminates a Call independent signalling connection

TETRA Subscriber Identity (TSI): global TETRA network address that is to identify an individual or a group subscriber within the domain of all TETRA networks

user: entity using the services of a telecommunications network via an externally accessible service access point

NOTE: An individual user may be a person or an application process.

user information: TETRA coded speech

visited SwMI: TETRA network which MNI is not equal to the user's MNI

3.2   Symbols

Void.

3.3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ANF	Additional Network Feature
ANF-ISI	all Additional Network Features of the Inter-System Interface
AVP	Audio Video Profile
C	Conditional
GFP	Generic Functional Protocol
GTSI	Group TETRA Subscriber Identity
IC	Individual Call
IP	Internet Protocol
ISI	Inter-System Interface
ISIGC	Inter System Interface Group Call
ISIIC	Inter System Interface Individual Call
ISIMM	Inter System Interface Mobility Management
ISISDS	Inter System Interface Short Data Service
ITSI	Individual TETRA Subscriber Identity
LA	Location Area
M	Mandatory
MCC	Mobile Country Code
MNC	Mobile Network Code
MNI	Mobile Network Identity
MRS	Mobile Radio Stack
MS	Mobile Station
O	Optional
PDU	Protocol Data Unit
RFC	Request For Comments
RTP	Real-time Transport Protocol
SDL	Specification and Description Language
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SSI	Short Subscriber Identity
SwMI	TETRA Switching and Management Infrastructure
TCP	Transmission Control Protocol
TLS	Transport Layer Security
TSI	TETRA Subscriber Identity
UDP	User Datagram Protocol
URI	Uniform Resource Identifier
V+D	Voice plus Data

4 Usage of Session Initiation Protocol (SIP) and Session Description Protocol (SDP) for TETRA

4.1 SIP functionality

The TETRA ISI application can use the Session Initiation Protocol (SIP) as transport mechanism for call control information between interconnected SwMIs. When using SIP as transport for call control information, Session Description Protocol (SDP) shall be used for the specification of the audio connection between interconnected SwMIs.

TETRA uses the SIP standard functionalities as defined in the following RFCs:

- IETF RFC 2045 [17].

- IETF RFC 3261 [8]. The following SIP request methods are used:

    - OPTIONS.

    - INVITE.

- ACK.

- CANCEL.

- BYE.

- IETF RFC 3428 [10].

- IETF RFC 4028 [11].

- IETF RFC 5621 [13].

- IETF RFC 6026 [14].

- IETF RFC 6086 [15].

- IETF RFC 6141 [16].

NOTE 1: Provisional responses to SIP:INVITE except 100 Trying are not used.

NOTE 2: The status of the SIP session is independent of the status of the ISI session e.g. the SIP session can be fully established (SIP:200 OK and SIP:ACK has been exchanged) while the ISI session is during establishment.

TETRA uses the SDP standard functionalities as defined in the following RFC:

- IETF RFC 4566 [12].

- IETF RFC 3264 [9].

SIP messages convey both TETRA ISI PDUs and SDP PDUs in the body part of the SIP messages.

4.2   Protocol stack for call control signalling information

Figure 4.1 shows how the protocols for TETRA ANF which apply at the ISI using SIP as transport mechanism.
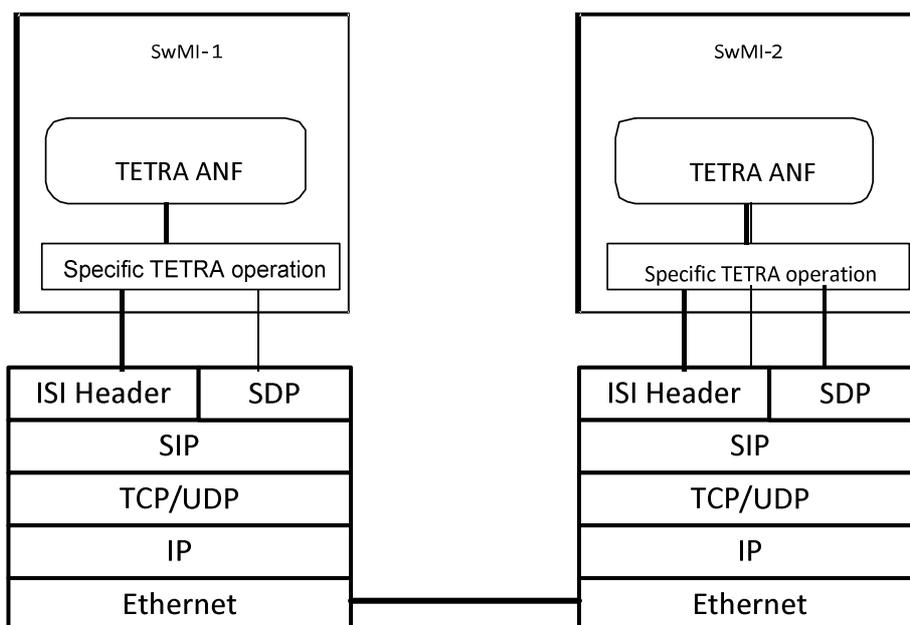
NOTE: Layer 2 protocol depends on the type of the inter-connection.

Figure 4.1: IP protocol stack for TETRA

The ISI header is used to convey ANF-ISI Protocol Data Units (PDU) to and from the correct TETRA ANF sessions. The ISI header identification is the same for all messages related to the same TETRA ANF session.

SIP can be conveyed between SwMIs using TLS, TCP or UDP. For security reasons it is recommended that the path between the SwMIs is encrypted.

4.3   Protocol stack for user information

Call control manages transportation of user information e.g. TETRA coded speech on RTP as presented in figure 4.2.

Figure 4.2: User information protocol stack

For speech transmission the TETRA coded user information is speech frames that shall be carried in RTP frames, refer to ETSI EN 300 392-3-8 [2]. The call control negotiates the ports for each call instance.
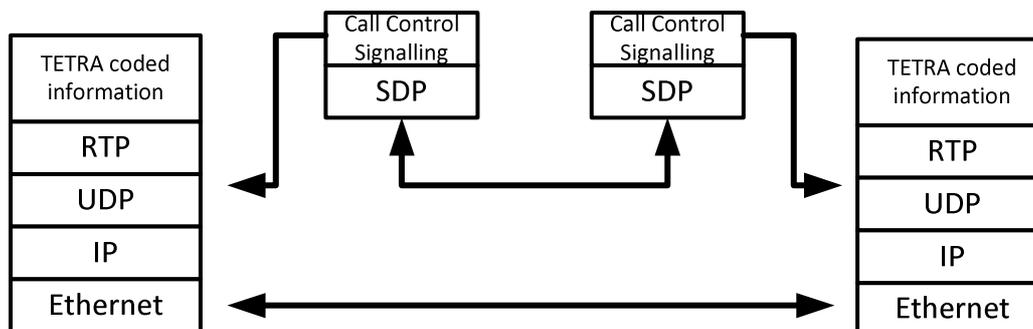
RTP is conveyed between the SwMIs using UDP. For security reasons it is recommended that the path between the SwMIs is encrypted.

5   ISI Generic Functional Protocol (ISI GFP)

5.1   Protocol model

The TETRA ISI over IP is based on the SIP protocols stack in order to enable the interconnection of SwMIs. Signalling needs for TETRA ISI operation which are not directly supported by SIP protocols are provided by ISI Mediation Function.

ISI Generic Function does not by itself control any ANF-ISI PDUs but rather provides a means to convey them.

Figure 5.1 shows the conceptual model of the ISI Generic Function for TETRA ISI over IP and its relationship to the various ANF entities. The generic ISI Generic Function is comprising the ISI Mediation Function including the PDU Identification Service which identifies the ANF, the session and the Audio Path Service.
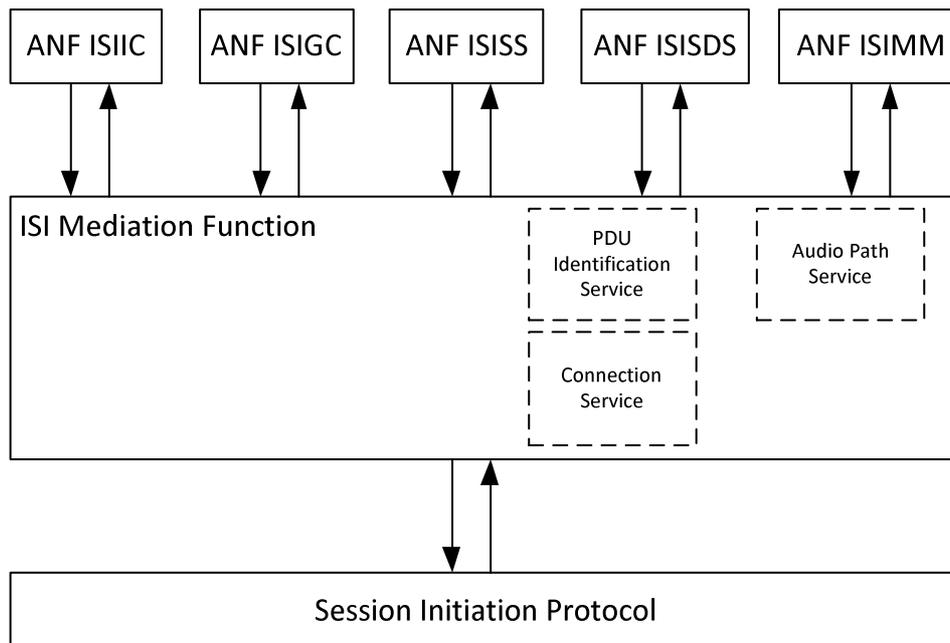
Figure 5.1: Protocol model of the ISI Generic Function Model

The ISI Mediation function coordinates the identification of message flows and controls the conveying of the message flows across the SIP protocol. The ISI Mediation Function for ISI over IP also controls the audio connection between the SwMIs by generating and interpreting SDP information conveyed in the SIP message body.

The entities ISI Mediation Function and Session Initiation Protocol shall exist in any SwMI which supports the ISI over IP. The existence in an SwMI of the other entities shown on figure 5.1 shall depend on the ANF-ISIs supported by this SwMI.

5.2   Services provided by the ISI Mediation Function

5.2.1   ISI Mediation Function Overview

The ISI Mediation Function provides the following services for the ANFs:

- Order establishment and release of relations between SwMIs based on the ISI signalling received from ANFs.

- Negotiate conditions for the audio connection between SwMIs (SDP).

- Coordination of ISI Invoke Identifiers between ANFs so that an ISI Invoke identifier is unique for a given time.

- Creation of the ISI header for sent ANF PDUs.

- ISI header validation for received messages:

  - This includes sending ISI Return Error messages for received messages where the header format is malformed or incomplete.

- Sending ISI Reject when an ISI Invoke PDU is addressed to an ANF-ISI entity which is not present/supported in the receiving SwMI.

- Sending ISI Return Error messages on behalf of the ANFs when the TETRA PDU is malformed or incomplete.

5.2.2   Session Description Protocol

5.2.2.1   General

The Session Description protocol shall only be invoked for streaming related ANF services e.g. ISI-IC and ISI-GC. The Session Description Protocol (SDP) is used to negotiate the media on which the streaming shall flow between two SwMIs. The SDP is described in IETF RFC 4566 [12].

The ISI Mediation Function shall support the SDP Offer/Answer model as described in IETF RFC 3264 [9].

Manufacturers may choose to offer vendor specific to provide the possibility for using proprietary codecs when two SwMIs from the same manufacturer are connected.

Following clauses describe the usage of mandatory lines of an SDP offer and answer.

5.2.2.2   Protocol Version ("v=")

See IETF RFC 4566 [12].

5.2.2.3   Origin ("o=")

See IETF RFC 4566 [12].

5.2.2.4   Session Name ("s=")

Session Name shall include the 'ISI Invoke ID' (see clause 7.1) in order to connect the media stream to the ANF PDUs.

5.2.2.5   Connection Data ("c=")

See IETF RFC 4566 [12].

5.2.2.6   Timing ("t=")

See IETF RFC 3264 [9].

5.2.2.7   Media Type ("m=") and associated Attributes ("a=")

The Media Type field is defined as "m=<media> <port> <proto/number of ports> <fmt>" where the:

- <media> shall be specified as 'audio'.

- <proto/number of ports> shall be a single port.

- <proto> shall be specified as 'RTP/AVP'.

- <fmt> shall contain at least one media format description.
  The <fmt> may contain more than one media format description, but at least one of them shall have the attributes equal to TETRA_ACELP.

The following media attributes shall be supported:

- a=rtpmap   (for each fmt in the m-line)

- a=sendrecv   (the attribute may be omitted, but shall be handled as the only supported value)

- a=ptime:60   (the attribute may be omitted, but shall be handled as the only supported value)

- a=maxptime:60   (the attribute may be omitted, but shall be handled as the only supported value)

Other attributes may be ignored if they are included in the SDP content.

5.2.2.8   Example of SDP Content

The following headers fields are used for negotiating the streaming between two SwMIs.

Table 5.1: SDP content

```
[offer conveyed in SIP INVITE]
Content-Type: application/sdp
v=0
o=- 234567890 234567890 IN IP4 151.3.7.100
s=549755813888
c=IN IP4 151.3.7.200
t=0 0
m=audio 45678 RTP/AVP 119 120 121
a=rtpmap:119 TETRA_ACELP/8000
a=rtpmap:120 x.vnd.some-TETRA-vendor.codec/8000
a=rtpmap:121 x.private.codec/8000
a=ptime:60
a=maxptime:60
a=sendrecv

[answer conveyed in SIP 200 OK]
Content-Type: application/sdp
v=0
o=- 234568901 234568901 IN IP4 87.34.5.300
s=549755813888
c=IN IP4 87.35.5.41
t=0 0
m=audio 56789 RTP/AVP 119
a=rtpmap:119 TETRA_ACELP/8000
a=ptime:60
a=maxptime:60
a=sendrecv
```

6   Session Initiation Protocol (SIP)

6.1   SIP Introduction

SIP can be used as transport mechanism between two SwMIs.

When a call is initiated between two SwMIs a SIP session is established. The session can be used for the specific call only. When the call is terminated then the SIP session shall be terminated as well.

For call independent signalling such as a mobility management sequence or an authentication sequence, a SIP session will also be established. The SIP session shall be terminated when the mobility management sequence or authentication sequence is finalized.

When the SIP session is initialized, the dialog is fully established immediately without waiting for the ANF to respond to the received ISI PDU. That means that SIP:200 OK is sent immediately independent on the content and the readability of the ISI PDU (see figure 6.1 and figure 6.6). This way the SIP session status is independent of the ANF session status.

For short data PDUs or status PDUs a session is not established. For these PDUs a SIP:MESSAGE is used to convey the ISI PDU.

6.2   SIP methods

The following SIP methods are to be supported for transport of ISI PDUs:

- OPTIONS, 200 OK.

- INVITE, 200 OK, ACK.

- INFO, 200 OK.

- BYE, 200 OK.

- MESSAGE, 200 OK.

NOTE 1: Provisional responses to SIP:INVITE is not to be supported. Any received provisional response should be ignored.

NOTE 2: The use of mid-dialog OPTIONS is not expected as the capability of the two SwMIs is expected to be stable and equal for all calls.

The following responses may be sent as a non-positive response to SIP:INVITE:

- 400 Bad Request:

  - May be used in case the content of the SIP header fields is corrupted.

- 403 Forbidden:

  - To be used if any other methods than the above mentioned are used or if a request is received from an unknown SwMI.

- 415 Unsupported MediaType:

  - To be used if message body content is not either SDP or ISI information.

- 416 Unsupported URI Scheme:

  - To be used when the Request URI is unknown.

- 422 Session Interval Too Small:

  - To be sent if the Session-Expires header field has a lower value than the minimum session timer supported by the terminating SwMI.

- 480 Temporary Unavailable:

  - May be used by an intermediate gateway if it does not have any resources, e.g. an IP2E1 gateway does not have any E1 links available (all line disconnected).

- 481 Call/Transaction does not exist:

  - To be used when no active call correspond to the Call-ID.

- 486 Busy Here:

  - May be used by an intermediate gateway if available resources are used, e.g. an IP2E1 gateway.

- 487 Request Terminated:

  - Used when a request method is received for an already terminated call.

- 500 Server Internal Failure:

  - The use is operator specific.

- 503 Service Unavailable:

  - To be used if the server is overloaded (too many requests compared to immediate processing capacity e.g. too many simultaneous call attempts).

- 600 Busy Everywhere:

  - To be used if the server is overloaded (too many requests compared to overall capacity e.g. too many simultaneous calls).

The following responses may be sent for a non-positive response to SIP:INFO:

- 400 Bad Request:

    - May be used in case the content of the SIP header fields is corrupted.

- 415 Unsupported MediaType:

    - To be used if message body content is not either SDP or ISI information.

- 481 Call/Transaction does not exist:

    - To be used when no active call correspond to the Call-ID.

- 487 Request Terminated:

    - Used when the a request method is received for an already terminated call.

- 500 Server Internal Failure:

    - The use is operator specific.

The following responses may be sent for a non-positive response to SIP:MESSAGE:

- 400 Bad Request:

    - May be used in case the content of the SIP header fields is corrupted.

- 401 Unauthorized:

    - May be used in case the requesting SwMI is unknown.

- 403 Forbidden:

    - To be used if any other methods than the above mentioned are used.

- 415 Unsupported MediaType:

    - To be used if message body content is not either SDP or ISI information.

- 416 Unsupported URI Scheme:

    - To be used when the Request URI is unknown.

- 480 Temporary Unavailable:

    - May be used by an intermediate gateway if it does not have any resources, e.g. an IP2E1 gateway does not have any E1 links available (all line disconnected).

- 486 Busy Here:

    - May be used by an intermediate gateway if available resources are used, e.g. an IP2E1 gateway.

- 500 Server Internal Failure:

    - The use is operator specific.

- 503 Service Unavailable:

    - To be used if the server is overloaded (too many requests).

- 600 Busy Everywhere:

    - To be used if the server is overloaded (too many requests).

The following responses may be sent for a non-positive response to SIP:OPTIONS:

- 400 Bad Request:

    - May be used in case the content of the SIP header fields is corrupted.

- 401 Unauthorized:

    - May be used in case the requesting SwMI is unknown.

- 416 Unsupported URI Scheme:

    - To be used when the Request URI is unknown.

- 480 Temporary Unavailable:

    - May be used by an intermediate gateway if it does not have any resources, e.g. an IP2E1 gateway does not have any E1 links available (all line disconnected).

- 486 Busy Here:

    - May be used by an intermediate gateway if available resources are used, e.g. an IP2E1 gateway.

- 500 Server Internal Failure:

    - The use is operator specific.

- 503 Service Unavailable:

    - To be used if the server is overloaded (too many requests).

- 600 Busy Everywhere:

    - To be used if the server is overloaded (too many requests).

The requesting SwMI shall be prepared for all possible response codes as proxy servers might send other response codes than those mentioned above.

6.3 SIP header fields

6.3.1 SIP header fields overview

This clause will list and describe the header fields which are mandatory for ISI communication. Additional header fields may be added to a SIP message. If an additional header field is present but not understood by the receiver of the message, the header field and its content shall be ignored. It is not allowed to reject a SIP message due to unexpected header fields.

If a header field among those mentioned in the present document has an additional content which is not described in the present document and the receiver of the message does not expect or understand the content the SIP message, the content shall be ignored. If a header field is empty after not expected/understood content is ignored it is up to the receiver to determine if the message can be accepted.

The general rules for the content of SIP header fields are described in IETF RFC 3261 [8]. The following clauses will only describe how the header fields are used in connection to ISI.

NOTE: The header fields do not contain any ANF specific information.

6.3.2 To

The To header field shall contain the SIP URI of the called SwMI. See clause 6.5.1 for the addressing of a SwMI.

The content of the To header field shall always correspond to the called SwMI, and never to any TETRA subscriber.

6.3.3   From

The From header field shall contain the SIP URI of the calling SwMI. See clause 6.5.1 for the addressing of a SwMI.

The From header field shall always correspond to requesting SwMI, and never to any TETRA subscriber.

6.3.4   Via

The Via header field shall initially contain the IP address of the calling SwMI. See clause 6.5.1 for the addressing of a SwMI.

6.3.5   Call-ID

See IETF RFC 3261 [8].

6.3.6   CSeq

See IETF RFC 3261 [8].

6.3.7   Max-Forwards

See IETF RFC 3261 [8].

6.3.8   Contact

The Contact header field shall contain the SIP URI of the calling SwMI. See clause 6.5.1 for the addressing of a SwMI.

The Contact header field shall always correspond to requesting SwMI, and never to any TETRA subscriber.

6.3.9   Content-Type

The Content-Type header field is used to indicate the content of the message body. If the message body contains more than one body part the first Content-Type header field shall indicate this and define boundary between the body parts in the following way:

```
Content-Type: multipart/mixed; boundary=tetra
.... ...
.... ...
--tetra
Content-Type: application/sdp
    <sdp content>
--tetra
Content-Type: application/TETRA_ISI
    <TETRA_ISI information>
--tetra--
```

6.3.10   Content-Disposition

The Content-Disposition header field describes how the message body or, in the case of multipart messages, a message body part is to be interpreted by the receiving SwMI.

The Content-Disposition header field for transport of ISI PDUs is only used in SIP:INFO to indicate that the type of message in the body part is belonging to an Info-Package.

```
Content-Disposition: Info-Package
```

   NOTE: The Info-Package header field indicates which package that the SIP:INFO transport. In this case the package is 'TETRA_ISI'.

6.3.11 Content-Length

See IETF RFC 3261 [8].

6.3.12 Allow

The Allow header field is not present in a request it is expected that the requesting SwMI supports any SIP method. If the Allow header field is present in a request message it shall contain at least the following information.

EXAMPLE 1: Within a specific SIP dialog the following requests are supported:

SIP:INVITE

Allow: INVITE, BYE, ACK, CANCEL, INFO

EXAMPLE 2: A SwMI can request information about all supported SIP methods by sending SIP:OPTIONS. In the response the requested SwMI shall respond with SIP:200 OK with at least the following content:

200 OK (OPTIONS)

Allow: INVITE, BYE, ACK, CANCEL, INFO, MESSAGE

6.3.13 Accept

The Accept header field in SIP:INVITE and SIP:200 OK (INVITE) shall contain at least the following applications:

- sdp.

- TETRA_ISI.

The application names are not case sensitive. The application information might be sent in separate SIP messages or all application information is sent in one SIP message.

EXAMPLE: Accept: application/sdp,application/TETRA_ISI.

6.3.14 Recv-Info

The Recv-Info header field shall indicated that TETRA_ISI Info Packages are accepted in received SIP:INFO requests. The header is present in SIP:INVITE and SIP:200 OK(INVITE).

EXAMPLE: Recv-Info: TETRA_ISI.

6.3.15 Info-Package

The Info-Package header field specifies the Info Package name. The Info-Package name for ISI communication is 'TETRA_ISI'. The header is shall always present in the SIP:INFO.

EXAMPLE: Info-Package: TETRA_ISI.

6.3.16 MIME-Version

The MIME-Version header fields specifies the MIME-Version used in the body part of the SIP message. It is required when more than one body part is present in the same SIP message.

EXAMPLE: MIME-Version: 1.0.

# 6.4		Availability and capability

## 6.4.1		Availability

SIP:OPTIONS may be used to determine if the peer SwMI/ISI application is available. It is recommended that SIP:OPTIONS is sent no more often than every 4 to 10 seconds. If the SIP:200 OK to SIP:OPTIONS is not received for a number of consecutive SIP:OPTIONS the peer SwMI/ISI application can be recognized as out of service and local service may be applied until the SIP:OPTIONS is responded. Each SwMI determines the action in case the peer SwMI is not responding.

NOTE:	If TCP is used instead of UDP it is possible to recognize faster if a link to the peer SwMI has failed, but it might not detect if the peer application is down.

## 6.4.2		Capability

The SIP:OPTIONS may be used to determine the capability of the connected SwMI. It is expected that each gateway supports all the RFCs listed in clause 4.1 and the SDP capability described in clause 5.2.2. In addition it is possible in SIP:200 OK to SIP:OPTIONS to indicate the capacity which can be supported. The following capacity parameters may be indicated:

```
Context-type:text/plain;charset=iso-8859-1
Number of sustained calls = xx
Number of new group calls per second  = yy
Number of new individual calls per second = zz
Number of migrating MSs per second = vv
Number of short data or status messages per second = ww
```

Zero, one or several of the capacity information lines may be included.

Receiving capacity information from the peer SwMI makes it possible for a SwMI to ensure that it is not overloaded the peer SwMI during operation. It is up to the receiver of the information if the capacity information of the peer SwMI is taken into account during operation.

# 6.5		Addressing and transport

## 6.5.1		Addressing

All SIP address header fields and SDP address shall be populated with the dotted-decimal representation of the IP version 4 globally unique address.

The header fields defined above shall always correspond to SwMIs or SwMI entities, and never to any TETRA subscriber.

## 6.5.2		Transport of ISI PDUs

ISI PDUs shall be encapsulated in the SIP message body. ISI PDUs are composed of arbitrary binary data that is transparent to SIP processing. The content of the SIP message is identified by a media type. The media type for ISI is defined by the following information:

- Media type name: application.

- Media subtype name: ISI.

- Required parameters: none.

- Optional parameters: none.

- Encoding scheme: binary.

The ISI PDU is binary coded as described in ETSI EN 300 392-3-9 [3], clause A.1. The encapsulated is done as described in IETF RFC 5621 [13] Message Body Handling in SIP.

6.6   Procedures

6.6.1   General

Only a single ISI PDU can be encapsulated in the SIP message body. If two ISI PDUs are to be sent right after each other, they shall be included in each their SIP message.

If the first of two ISI PDUs in a row initiates establishment of the session between two SwMIs the second ISI PDU cannot be sent before SIP:200 OK has been received and the SIP:ACK is sent for the initial SIP message.

If the two ISI PDUs are to be sent within an established connection then they shall be sent in separate SIP:INFO messages. It is recommended to wait for SIP:200 OK for the first SIP:INFO before the second SIP:INFO is sent. This way the order of the ISI PDUs can be ensured in any kind of transport network and independent of any re-transmission.

6.6.2   Call/Session Establishment

The procedures to establish a signalling connection shall be those defined in IETF RFC 3261 [8].

The connection is established immediately when the SIP:INVITE is received - before the ISI PDU is validated and processed. No ISI PDU can be included in the SIP:200 OK or SIP:ACK.
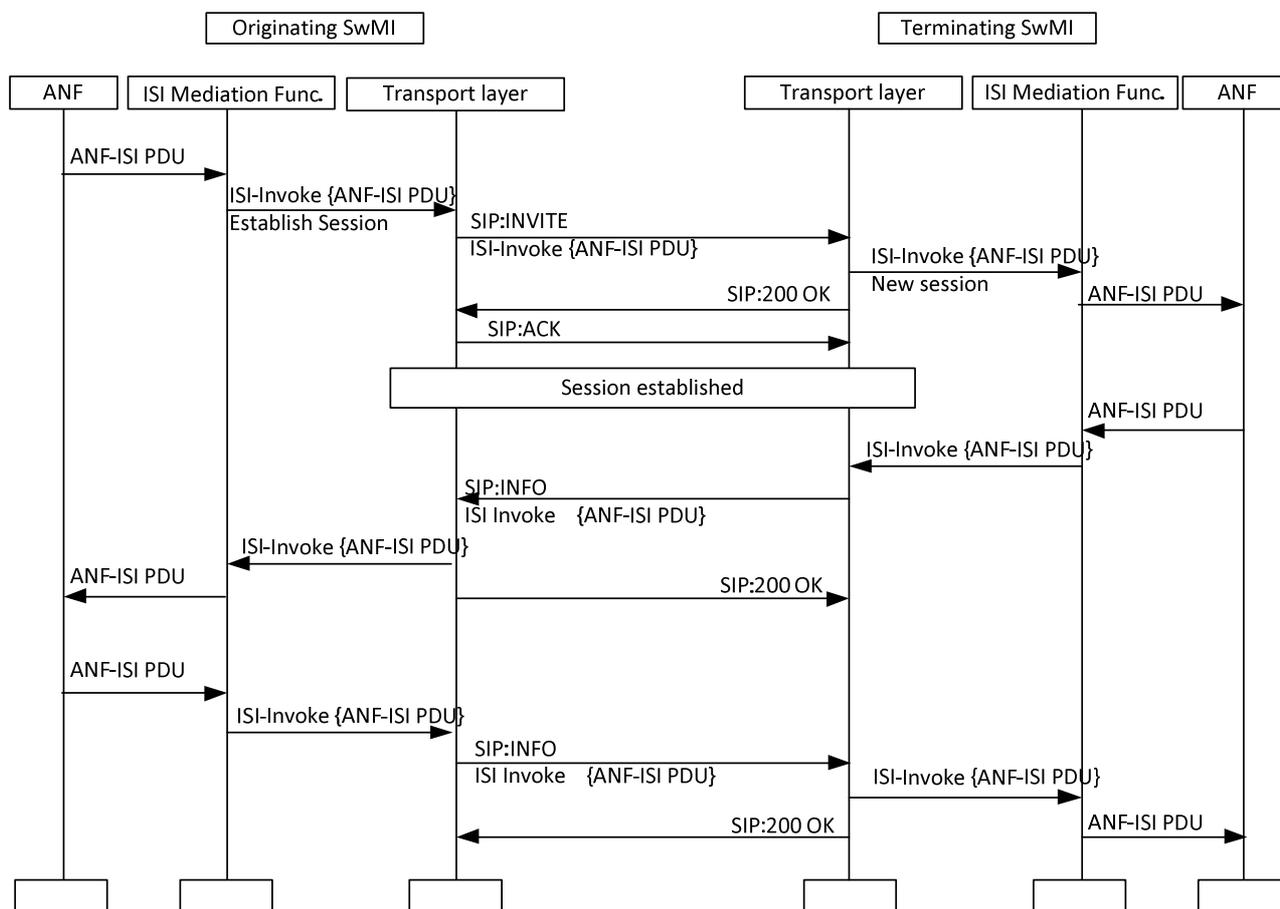

Figure 6.1: Establishment of signalling connection without audio connection

The transport layer shall not act on the content of the ISI PDU, but solely on orders from the ISI Mediation Function. For ANF-ISIMM according to ETSI EN 300 392-3-15 [7] and ANF-ISISDS according to ETSI EN 300 392-3-14 [6] PDUs no audio path is established. SIP:INVITE does not contain any SDP information the Transport layer can immediately acknowledge with SIP:200 OK to SIP:INVITE.

All subsequent ISI PDUs are sent in SIP:INFO, which is responded immediately. No ISI PDU can be included in SIP:200 OK to SIP:INFO.

For individual calls (as specified in ETSI EN 300 392-3-12 [4]) the audio connection is to be requested in relation to sending of the initial call establishment PDU.
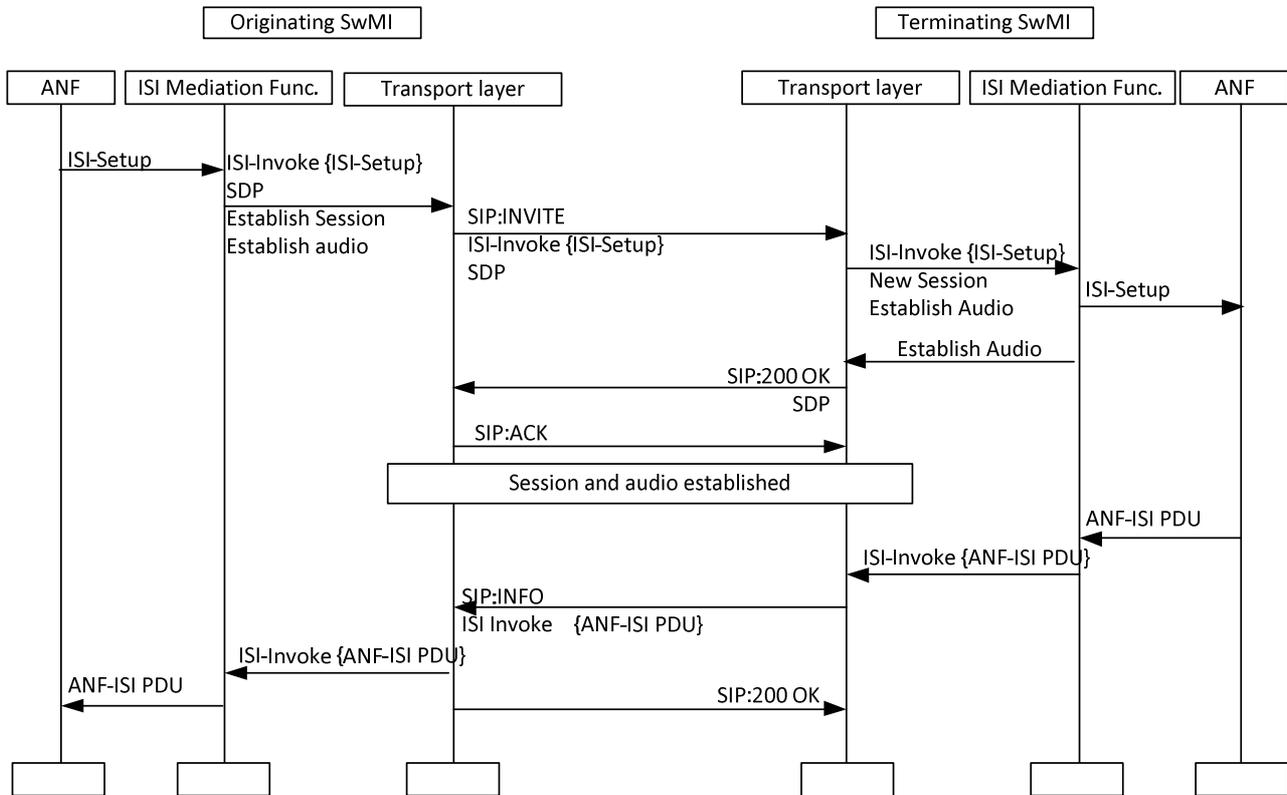

Figure 6.2: Establishment of an Individual Call

For group call initiation (as specified in ETSI EN 300 392-3-13 [5]) from an originating/participating SwMI the originating/participating SwMI ANF may not order an audio connection in connection to the group call request, but leave it to the controlling SwMI ANF to determine if the call shall be established. This brings the controlling SwMI ANF in total control of the call and the audio path and can prevent fraudulent audio establishment between the SwMIs. This means that the session may be established without exchange of SDP information. When the controlling SwMI ANF has determined that the call shall be established, it will request the establishment of the call and the audio path in a Re-INVITE including an ANF-ISI PDU to establish the call and the SDP PDU to establish the audio connection.
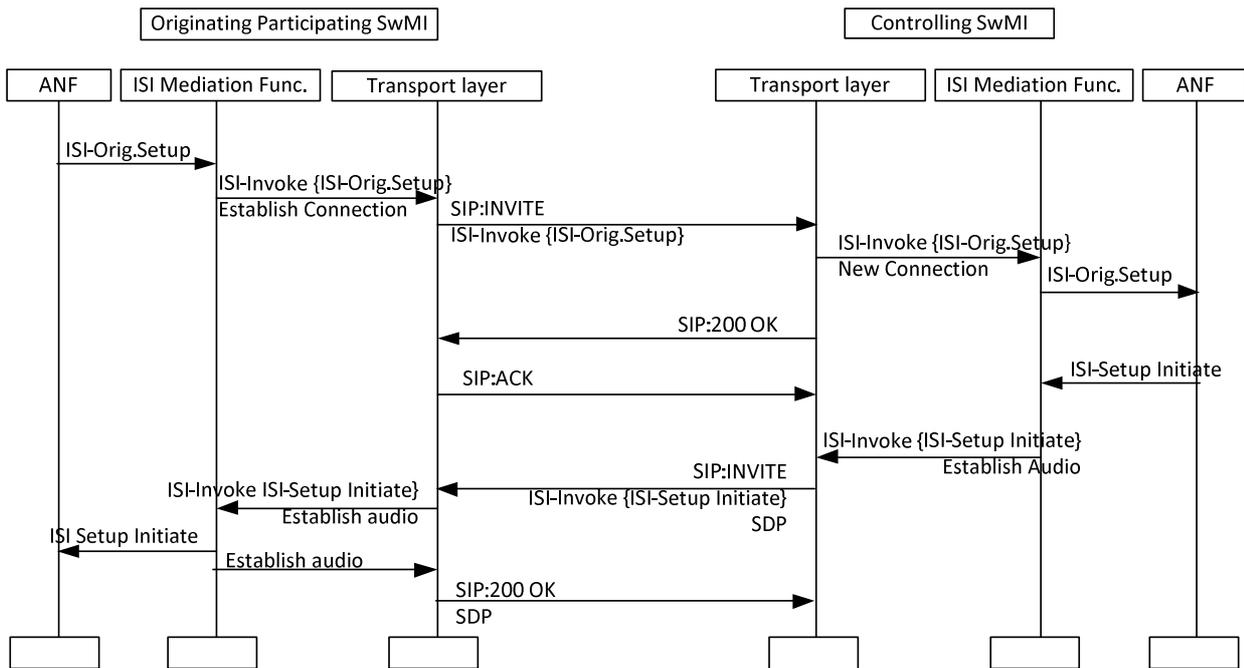
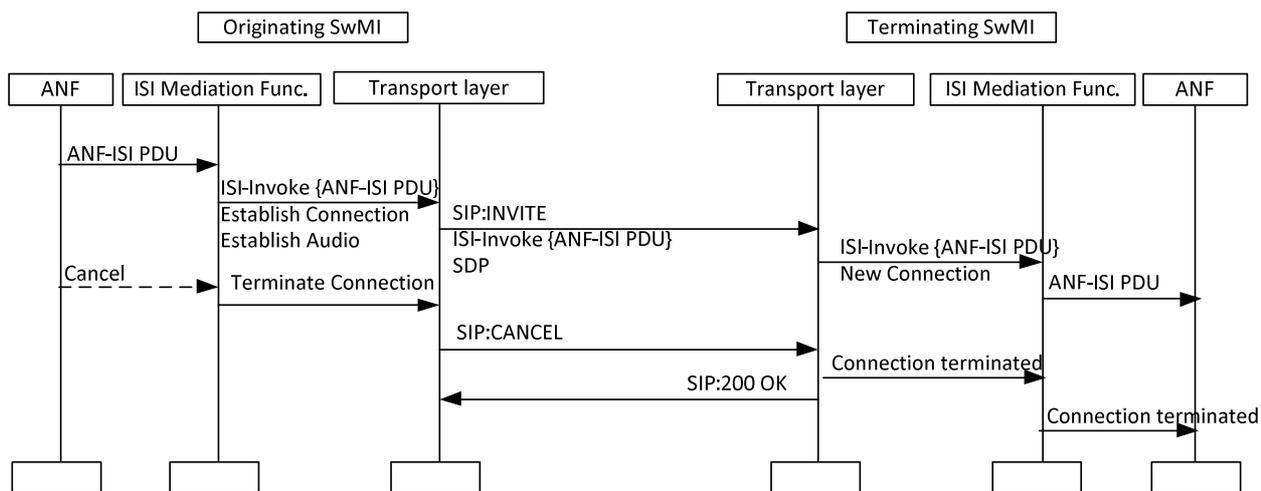**Figure 6.3: Initiation of a Group Call from P-SwMI**

If the group call originates from the controlling SwMI the SwMI will establish the audio connection in relation to the establishment of the SIP connection.
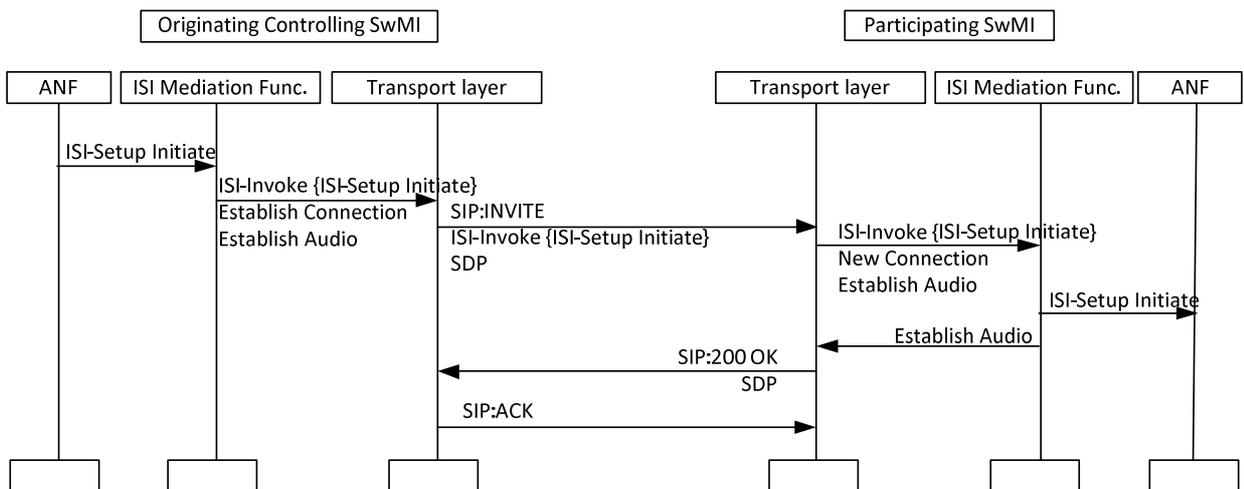
**Figure 6.4: Establishment of a Group Call from C-SwMI**

All other ANF-ISI PDUs except the final release ANF-ISI PDU of the call are sent in SIP:INFO.

If either the controlling or the originating SwMI decides to clear the call related signalling connection during call establishment, it shall do so according IETF RFC 3261 [8]. If the originating SwMI determines to terminate the call before the initial SIP:INVITE has been responded this is done by sending SIP:CANCEL.

Figure 6.5: Cancellation of a Group Call during Call Initiation

If the terminating SwMI ANF determines -for whatever reason - to reject the call request, it shall indicate a reason to be included in the ISI Reject message. The ISI Mediation function may also initiate the ISI Reject if the ISI header is corrupted or the ISI Invoke ID is already in use. In any case the initial SIP:INVITE shall be responded positively with SIP:200 OK and the ISI Reject shall be included in either a SIP:INFO or a SIP:BYE message.
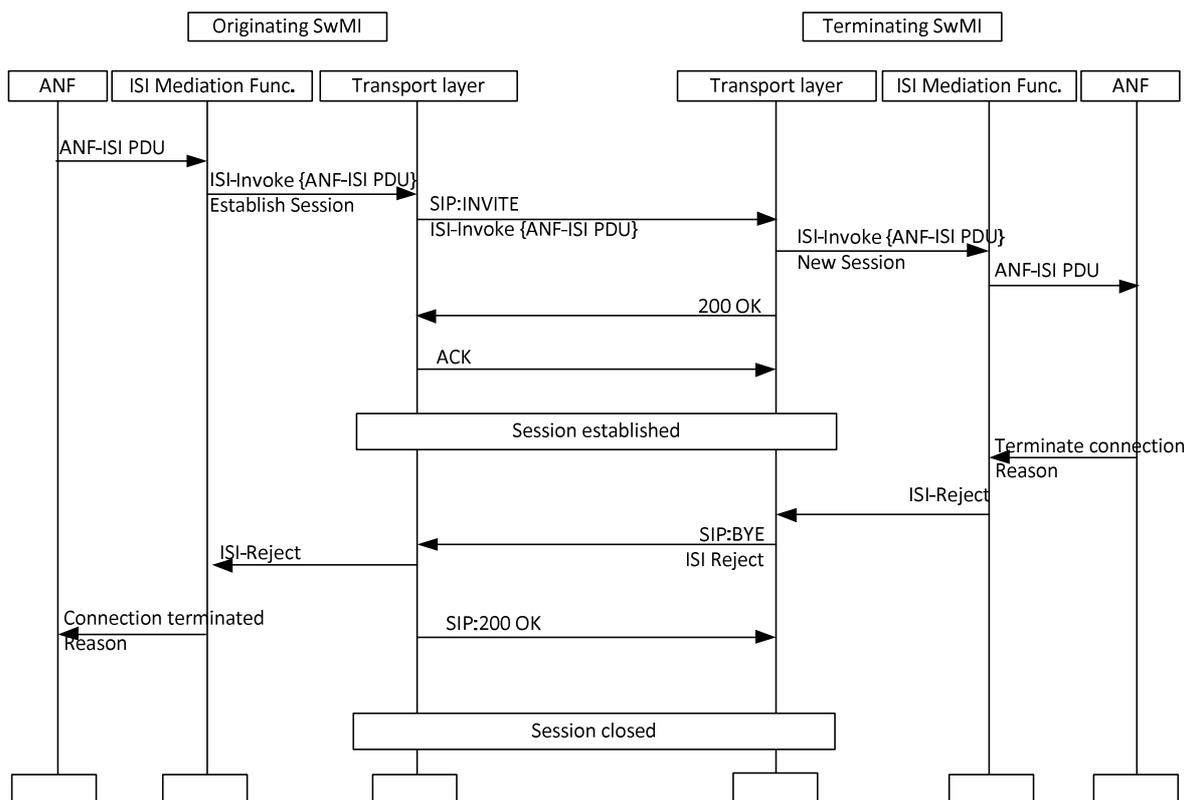
Figure 6.6: Error detected in the ANF during call establishment

NOTE: The ISI Reject may be sent in a SIP:INFO. In this case the dialog remains open. This gives the receiver of the ISI Reject or ISI Return Error an chance to correct the ISI PDU or to release the call.

6.6.3   Call/Session Termination

Normal termination may be initiated from any of the SwMIs. If more than one ANF-ISI PDU are exchanged during the termination only the very last ANF-ISI PDU in the sequence can be included in the SIP:BYE.

6.6.4   Message Sending

When sending Short Data or Status messages across ISI (as specified in ETSI EN 300 392-3-14 [6]) the SIP:MESSAGE is used. The response to SIP:MESSAGE indicates if the ISI PDU is successfully received in the terminating SwMI or not. It does not indicate if the content of the ISI PDU has been successfully delivered to the end user. Any response of that kind is sent in a new SIP:MESSAGE.

NOTE: In case a response to a short data or status message it is sent as a new independent PDU. The Invoke-IDs of the response ISI PDU are not coordinated with the forward PDU.
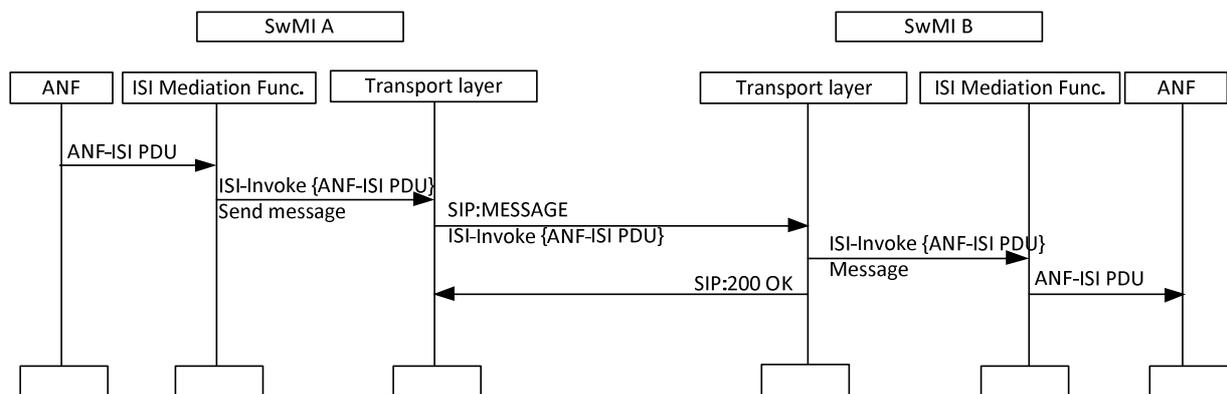
Figure 6.7: Sending of Short Data or Status Messages

6.7   Session Timer

6.7.1   Session Timer Overview

The requirements for the implementation of the session timer are described in IETF RFC 4028 [11].

The SIP session timer shall prevent hanging calls in case of link failure or dropped disconnect messages (SIP:BYE) along the path. The originating SwMI may request the session timer to be used for the call. The terminating SwMI may accept the use of the session timer. In that case it shall indicate to the originating SwMI who shall refresh the session before the session timer expires.

IETF RFC 4028 [11] recommends a session timer value of 1 800 seconds for normal calls. This value is based on a high number of available resources and a low number of failures. As the available resources for ISI usually is low (compared with telephony) it is recommended to use the minimum value (= 90 seconds) for the Session-Expires header field for communication between SwMIs.

6.7.2   Header fields for Session Timer

6.7.2.1   Supported

The Supported header field shall indicate if the Session Timer is supported.

EXAMPLE: Supported: timer.

6.7.2.2   Min-SE

The Min-SE header field indicates the minimum supported value of the session timer. IETF RFC 4028 [11] specifies that the minimum value is 90 seconds. It is recommended to use this value in the Min-SE header field for communication between SwMIs.

EXAMPLE: Min-SE: 90.

6.7.2.3 Session-Expires

The Session Expires header field indicates the requested value of the session timer.

EXAMPLE 1: Session-Expires in the initial SIP:INVITE:

Session-Expires: 90

EXAMPLE 2: Session-Expires in SIP:200 OK (INVITE):

Session-Expires: 90;refresher=aus

or

Session-Expires: 90;refresher=auc

EXAMPLE 3: Session-Expires in the refresh SIP:INVITE:

Session-Expires: 90;refresher=auc

6.7.3 Procedures

The procedure for the session timer negotiation is described in IETF RFC 4028 [11]. If the terminating SwMI supports the session timer functionality it determines who shall be the refresher. No further negotiation of timer values or refresher shall be performed.
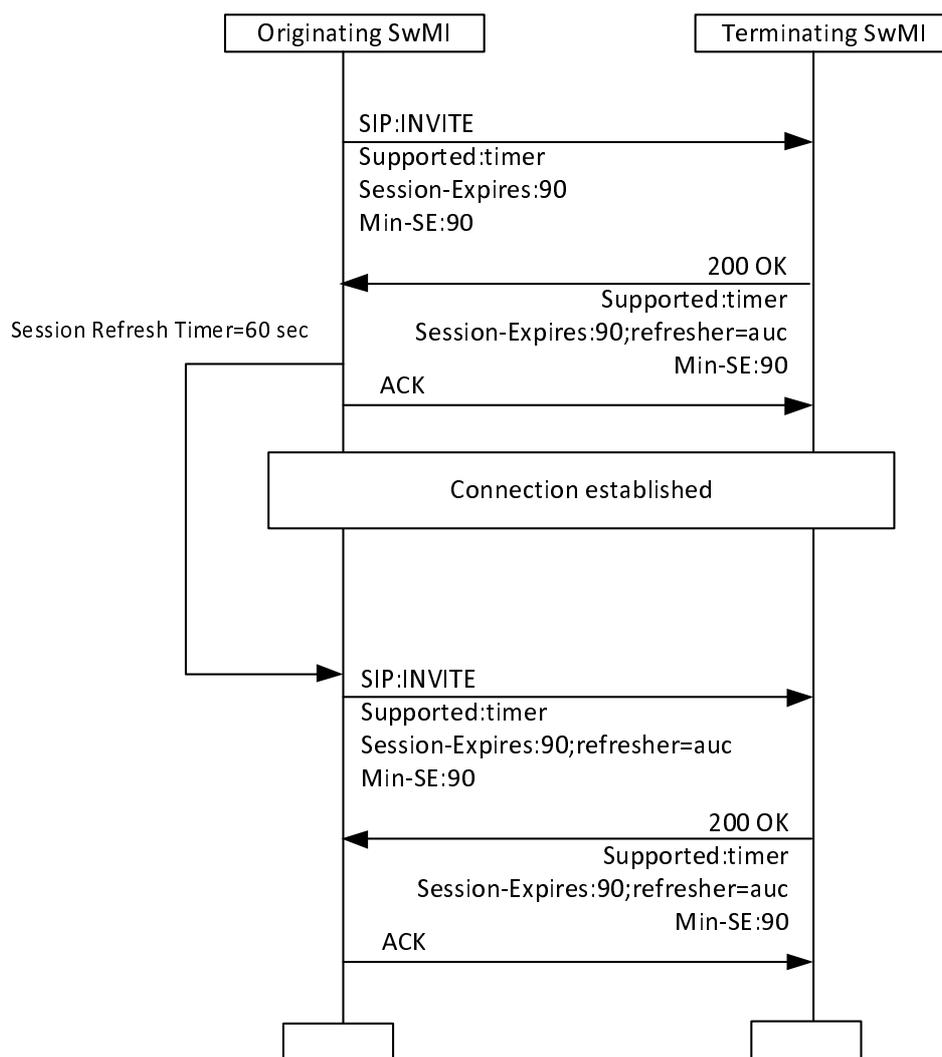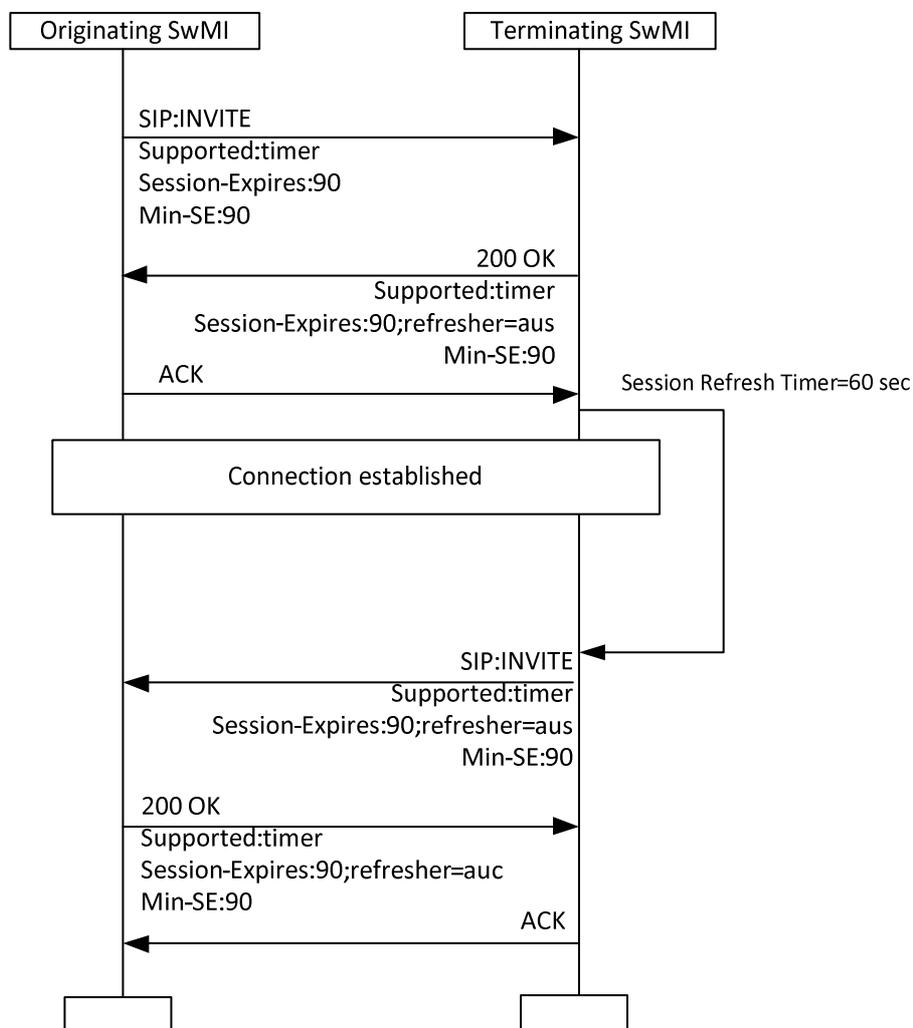

Figure 6.8: SIP session timer, Originating SwMI as refresher

The refresher SwMI shall start the refresh of the session after 60 seconds. This leaves room for one or more lost SIP refresh messages before the session timer expires.

The other SwMI shall start the session timer = 90 seconds. When a refresh SIP:INVITE is received then the session timer = 90 seconds shall be restarted. If no refresh has been received before the session timeout, the call shall be released by sending SIP:BYE towards the refreshing SwMI. The reason for sending SIP:BYE is to disconnect any intermediate security gateways.

Figure 6.9: SIP session timer, Terminating SwMI as refresher

If it happens that the refresher does not receive any response from the other SwMI, the SIP:INVITE will be repeated as per IETF RFC 3261 [8]. When a response is received the refresh timer shall be re-started.

If no response is received before the session timer has expired the refresher need to terminate the call by sending SIP:BYE towards the not responding SwMI. The reason for sending SIP:BYE is to disconnect any intermediate security gateways.

7 Special Considerations

7.1 ISI Invoke ID

The ISI Invoke ID shall be unique for a call at a given time. In order to avoid that the same ISI Invoke ID is used in two different calls initiated by either end of an ISI link it is recommended to precede the ISI Invoke ID with the MNI of the initiating SwMI and construct the INTEGER in 5 bytes in the following way:

ISI Invoke ID =[<mni><random within -32 768 to 32 767>]

Where <mni> is contained in 24 bits (3 bytes) and < random within -32 768 to 32 767> is contained in 2 bytes.

The construction of the ISI invoke D shall not be validated by the terminating SwMI.

Annex A (informative):
SIP Message Content

A.1   SIP basic call message

```
INVITE sip:user@192.168.0.200:5060 SIP/2.0
Via: SIP/2.0/TCP 151.3.7.100:5060;branch=z9hG4bKaddb1e4f8b2f
From: user <sip:user@151.3.7.100:5060>;tag=5-6506649a-00000001
To: user <sip:user@87.34.5.300:5060>
Contact: user <sip:user@151.3.7.100:5060>
Recv-Info: TETRA_ISI
Call-ID: 65066499093b66c81@151.3.7.100
Accept: application/sdp, application/TETRA_ISI, multipart/mixed
Supported: timer
Session-Expires: 90
Min-SE: 90
Allow: INVITE, BYE, ACK, CANCEL, INFO, OPTIONS
CSeq: 1 INVITE
Max-Forwards: 70
MIME-Version: 1.0
Content-Type: multipart/mixed; boundary="tetra"
Content-Length: <length>
--tetra
Content-Type: application/sdp
v=0
o=<user> <session-id> <session version> IN IP4 <IP2E1 Signalling IP address>
s=<ISI InvokeID>
c=IN IP4 <audio port number>
t=0 0
m=audio IP4 <audio IP address > RTP/AVP 119
a=rtpmap: 119 TETRA_ACELP/8000
a=sendrecv
a=ptime:60
a=maxptime:60
--tetra
Content-Type: application/ISI
A1 4A 02 02 0C C0 06 05 04 00 83 08 00 30 3D 80
01 05 81 01 05 82 35 00 07 92 89 D2 02 4A 84 8D
15 9E 24 00 3C 97 0F 40 12 94 13 0E CA 86 42 12
F1 91 04 00 F6 10 A6 88 40 DE EC CA E4 40 A8 CA
E8 E4 C2 40 92 A6 92 5E A6 92 A0 80
--tetra--
```

NOTE 1:   Since binary encoding is used for the ISI payload, each byte is encoded as a byte, and not as a two-character hex representation. Hex digits were used in the present document because a literal encoding of those bytes would have been confusing and unreadable.

NOTE 2:   For an example of the binary coding of the ISI PDU see ETSI EN 300 392-3-9 [3].

Annex B (informative):
Change requests

The present document includes change requests as presented in table B.1.

Table B.1: Change requests

No	CR vers.	Standard Version	Clauses affected	Title	CR Status
001	01	1.1.2	2.1, 4.3, 6.5.2, 6.6.2, 6.6.4, A.1	Correction to references	WG3 approved 190327

Annex C (informative):
Bibliography

- ETSI EN 300 392-7: "Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D); Part 7: Security".

- ETSI EN 300 392-9: "Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D); Part 9: General requirements for supplementary services".

- ETSI EN 300 395-1: "Terrestrial Trunked Radio (TETRA); Speech CODEC for full-rate traffic channel; Part 1: General description of speech functions".

- Recommendation ITU-T Z.100: "Specification and description language (SDL)".

History

Document history		
V1.1.1	May 2018	Publication as ETSI TS 100 392-3-11
V1.2.0	August 2019	EN Approval Procedure AP 20191113: 2019-08-15 to 2019-11-13
V1.2.1	April 2020	Publication