                Session Description Protocol (SDP) Extension for
     Setting Audio and Video Media Streams over Circuit-Switched Bearers in
                the Public Switched Telephone Network (PSTN)

Abstract

   This memo describes use cases, requirements, and protocol extensions
   for using the Session Description Protocol (SDP) offer/answer model
   for establishing audio and video media streams over circuit-switched
   bearers in the Public Switched Telephone Network (PSTN).

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1. Introduction

   The Session Description Protocol (SDP) [RFC4566] is intended for
   describing multimedia sessions for the purposes of session
   announcement, session invitation, and other forms of multimedia
   session initiation. SDP is most commonly used for describing media
   streams that are transported over the Real-Time Transport Protocol
   (RTP) [RFC3550], using the profiles for audio and video media defined
   in "RTP Profile for Audio and Video Conferences with Minimal Control"
   [RFC3551].

   However, SDP can be used to describe media transport protocols other
   than RTP. Previous work includes SDP conventions for describing ATM
   bearer connections [RFC3108] and the Message Session Relay Protocol
   [RFC4975].

   SDP is commonly carried in Session Initiation Protocol (SIP)
   [RFC3261] messages in order to agree on a common media description
   among the endpoints. "An Offer/Answer Model with the Session
   Description Protocol (SDP)" [RFC3264] defines a framework by which
   two endpoints can exchange SDP media descriptions and come to an
   agreement as to which media streams should be used, along with the
   media-related parameters.

   In some scenarios, it might be desirable to establish the media
   stream over a circuit-switched bearer connection even if the
   signaling for the session is carried over an IP bearer. An example
   of such a scenario is illustrated with two mobile devices capable of
   both circuit-switched and packet-switched communication over a low-
   bandwidth radio bearer. The radio bearer may not be suitable for
   carrying real-time audio or video media, and using a circuit-switched
   bearer would offer a better perceived quality of service. So,
   according to this scenario, SDP and its higher-layer session control
   protocol (e.g., the Session Initiation Protocol (SIP) [RFC3261]) are
   used over regular IP connectivity, while the audio or video is
   received through the classical circuit-switched bearer.

   This document addresses only the use of circuit-switched bearers in
   the PSTN, not a generic circuit-switched network. The mechanisms
   presented below require a call signaling protocol of the PSTN to be
   used (such as ITU-T Q.931 [ITU.Q931.1998] or 3GPP TS 24.008
   [TS.24.008]).

Setting up a signaling relationship in the IP domain instead of just
setting up a circuit-switched call also offers the possibility of
negotiating, in the same session, other IP-based media that is not
sensitive to jitter and delay, for example, text messaging or
presence information.

At a later point in time, the mobile device might move to an area
where a high-bandwidth packet-switched bearer, for example, a
Wireless Local Area Network (WLAN) connection, is available. At this
point, the mobile device may perform a handover and move the audio or
video media streams over to the high-speed bearer. This implies a
new exchange of SDP offer/answer that leads to a renegotiation of the
media streams.

Other use cases exist. For example, an endpoint might have at its
disposal circuit-switched and packet-switched connectivity, but the
same audio or video codecs are not feasible for both access networks.
For example, the circuit-switched audio or video stream supports
narrow-bandwidth codecs, while the packet-switched access allows any
other audio or video codec implemented in the endpoint. In this
case, it might be beneficial for the endpoint to describe different
codecs for each access type and get an agreement on the bearer
together with the remote endpoint.

There are additional use cases related to third party call control
where the session setup time is improved when the circuit-switched
bearer in the PSTN is described together with one or more codecs.

The rest of the document is structured as follows: Section 2 provides
the document conventions, Section 3 introduces the requirements,
Section 4 presents an overview of the proposed solutions, and
Section 5 contains the protocol description. Section 6 provides
examples of circuit-switched audio or video streams in SDP. Sections
7 and 8 contain the Security and IANA considerations, respectively.

2. Conventions Used in This Document

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT",
"SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and
"OPTIONAL" in this document are to be interpreted as described in BCP
14, RFC 2119 [RFC2119] and indicate requirement levels for compliant
implementations.

3. Requirements

   This section presents the general requirements that are specific for
   the audio or video media streams over circuit-switched bearers.

   REQ-1: A mechanism for endpoints to negotiate and agree on an audio
           or video media stream established over a circuit-switched
           bearer MUST be available.

   REQ-2: The mechanism MUST allow the endpoints to combine circuit-
           switched audio or video media streams with other
           complementary media streams, for example, text messaging.

   REQ-3: The mechanism MUST allow the endpoint to negotiate the
           direction of the circuit-switched bearer, i.e., which
           endpoint is active when initiating the circuit-switched
           bearer.

   REQ-4: The mechanism MUST be independent of the type of the circuit-
           switched access (e.g., Integrated Services Digital Network
           (ISDN), Global System for Mobile Communication (GSM), etc.)

   REQ-5: There MUST be a mechanism that helps an endpoint to correlate
           an incoming circuit-switched bearer with the one negotiated
           in SDP, as opposed to another incoming call that is not
           related to that. In case correlation by programmatic means
           is not possible, correlation may also be performed by the
           human user.

   REQ-6: It MUST be possible for endpoints to advertise different
           lists of audio or video codecs in the circuit-switched audio
           or video stream from those used in a packet-switched audio or
           video stream.

   REQ-7: It MUST be possible for endpoints to not advertise the list
           of available codecs for circuit-switched audio or video
           streams.

4. Overview of Operation

   The mechanism defined in this memo extends SDP [RFC4566] and allows
   describing an audio or video media stream established over a circuit-
   switched bearer. A new network type ("PSTN") and a new protocol type
   ("PSTN") are defined for the "c=" and "m=" lines to be able to
   describe a media stream over a circuit-switched bearer. These SDP
   extensions are described in Section 5.2. Since circuit-switched
   bearers are connection-oriented media streams, the mechanism reuses

   the connection-oriented extensions defined in RFC 4145 [RFC4145] to
   negotiate the active and passive sides of a connection setup. This
   is further described in Section 5.3.1.

4.1. Example Call Flow

   Consider the example presented in Figure 1. In this example,
   Endpoint A is located in an environment where it has access to both
   IP and circuit-switched bearers for communicating with other
   endpoints. Endpoint A decides that the circuit-switched bearer
   offers a better perceived quality of service for voice and issues an
   SDP offer containing the description of an audio media stream over a
   circuit-switched bearer.

```
    Endpoint A                           Endpoint B
       |  (1) SDP offer (PSTN audio)        |
       |----------------------------------->|
       |                                    |
       |  (2) SDP answer (PSTN audio)       |
       |<-----------------------------------|
       |                                    |
       |     PSTN call setup                |
       |<-----------------------------------|
       |                                    |
       |                                    |
       |<===== media over PSTN bearer =====>|
       |                                    |
```
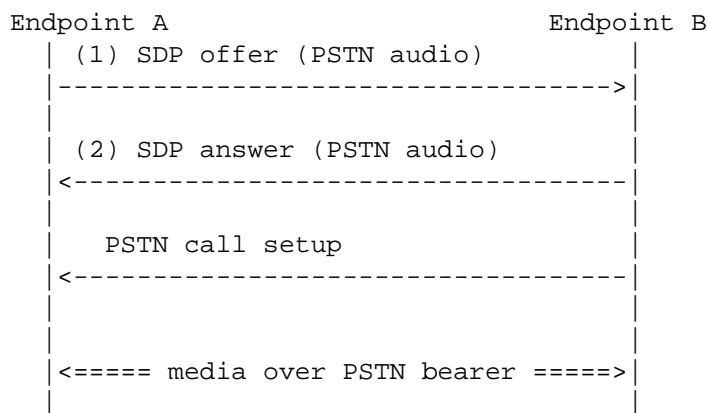
                    Figure 1: Example Flow

   Endpoint B receives the SDP offer and determines that it is located
   in an environment where the IP-based bearer is not suitable for real-
   time audio media. However, Endpoint B also has a PSTN circuit-
   switched bearer available for audio. Endpoint B generates an SDP
   answer containing a description of the audio media stream over a
   circuit-switched bearer.

   During the offer/answer exchange, Endpoints A and B also agree upon
   the direction in which the circuit-switched bearer should be
   established. In this example, Endpoint B becomes the active party;
   in other words, it establishes the circuit-switched call to the other
   endpoint. The offer/answer exchange contains identifiers or
   references that can be used on the circuit-switched network for
   addressing the other endpoint, as well as information that is used to
   determine that the incoming circuit-switched bearer establishment is
   related to the ongoing session between the two endpoints.

   Endpoint B establishes a circuit-switched bearer towards Endpoint A
   using whatever mechanisms are defined for the network type in
   question. When receiving the incoming circuit-switched connection
   attempt, Endpoint A is able to determine that the attempt is related
   to the session it is just establishing with B.

   Endpoint A accepts the circuit-switched connection; the circuit-
   switched bearer setup is completed. The two endpoints can now use
   the circuit-switched connection for two-way audio media.

   If, for some reason, Endpoint B would like to reject the offered
   stream, it would set the port number of the specific stream to zero,
   as specified in RFC 3264 [RFC3264]. Also, if B does not understand
   some of the SDP attributes specified in this document, it would
   ignore them, as specified in RFC 4566 [RFC4566].

5. Protocol Description

5.1. Level of Compliance

   Implementations that are compliant with this specification MUST
   implement the SDP extensions described in Section 5.2 and MUST
   implement the considerations discussed in Sections 5.3, 5.4, and 5.6.

5.2. Extensions to SDP

   This section provides the syntax and semantics of the extensions
   required for providing a description of audio or video media streams
   over circuit-switched bearers in SDP.

5.2.1. Connection Data

   According to SDP [RFC4566], the connection data line in SDP has the
   following syntax:

      c=<nettype> <addrtype> <connection-address>

   where <nettype> indicates the network type, <addrtype> indicates the
   address type, and <connection-address> is the connection address,
   which is dependent on the address type.

   At the moment, the only network type defined is "IN", which indicates
   Internet network type. The address types "IP4" and "IP6" indicate
   the type of IP addresses.

   This memo defines a new network type for describing a circuit-
   switched bearer network type in the PSTN. The mnemonic "PSTN" is
   used for this network type.

For the address type, we initially considered the possibility of
describing E.164 telephone numbers. We define a new "E164" address
type to be used within the context of a "PSTN" network type. The
"E164" address type indicates that the connection address contains an
E.164 number represented according to the ITU-T E.164 [ITU.E164.2010]
recommendation.

It is a common convention that an international E.164 number contains
a leading '+' sign. For consistency's sake, we also require the
E.164 telephone is prepended with a '+', even if that is not
necessary for routing of the call in the PSTN network.

There are cases, though, when the endpoint is merely aware of a
circuit-switched bearer, without having further information about the
E.164 number allocated to it. In these cases, a dash ("-") is used
to indicate an unknown connection address. This makes the connection
data line consistent with SDP syntax.

Please note that the "E164" address type defined in this memo is
exclusively defined to be used in conjunction with the "PSTN" network
type in accordance with regular offer/answer procedures [RFC4566].

   Note: RFC 3108 [RFC3108] also defines address type "E.164". This
   definition is distinct from the one defined by this memo and shall
   not be used with <nettype> "PSTN".

This memo exclusively uses the international representation of E.164
numbers, i.e., those including a country code and, as described
above, prepended with a '+' sign. Implementations conforming to this
specification and using the "E164" address type together with the
"PSTN" network type MUST use the 'global-number-digits' construction
specified in RFC 3966 [RFC3966] for representing international E.164
numbers. This representation requires the presence of the '+' sign
and additionally allows for the presence of one or more 'visual-
separator' constructions for easier human readability (see
Section 5.7).

Note that <connection-address> MUST NOT be omitted when unknown since
this would violate basic syntax of SDP [RFC4566]. In such cases, it
MUST be set to a "-".

The following are examples of the extension to the connection data
line:

   c=PSTN E164 +441134960123

   c=PSTN E164 -

When the <addrtype> is E164, the connection address is defined as
follows:

o   an international E.164 number (prepended with a '+' sign)

o   the value "-", signifying that the address is unknown

o   any other value resulting from the production rule of connection-
    address in RFC 4566 [RFC4566], but in all cases any value
    encountered will be ignored.

5.2.2. Media Descriptions

According to SDP [RFC4566], the media description line in SDP has the
following syntax:

    m=<media> <port> <proto> <fmt> ...

The <media> subfield carries the media type. For establishing an
audio bearer, the existing "audio" media type is used. For
establishing a video bearer, the existing "video" media type is used.

The <port> subfield is the transport port to which the media stream
is sent. Circuit-switched access lacks the concept of a port number;
therefore, the <port> subfield does not carry any meaningful value.
In order to be compliant with SDP syntax, implementations SHOULD set
the <port> subfield to the discard port value "9" and MUST ignore it
on reception.

According to RFC 3264 [RFC3264], a port number of zero in the offer
of a unicast stream indicates that the stream is offered but must not
be used. If a port number of zero is present in the answer of a
unicast stream, it indicates that the stream is rejected. These
rules are still valid when the media line in SDP represents a
circuit-switched bearer.

The <proto> subfield is the transport protocol. The circuit-switched
bearer uses whatever transport protocol it has available. This
subfield SHOULD be set to the mnemonic "PSTN" to be syntactically
correct with SDP [RFC4566] and to indicate the usage of circuit-
switched protocols in the PSTN.

The <fmt> subfield is the media format description. In the classical
usage of SDP to describe RTP-based media streams, when the <proto>
subfield is set to "RTP/AVP" or "RTP/SAVP", the <fmt> subfield
contains the payload types as defined in the RTP audio profile
[RFC3551].

When "RTP/AVP" is used in the <proto> field, the <fmt> subfield
contains the RTP payload type numbers. We use the <fmt> subfield to
indicate the list of available codecs over the circuit-switched
bearer, by reusing the conventions and payload type numbers defined
for RTP / AVP. The RTP audio and video media types, when applied to
PSTN circuit-switched bearers, represent merely an audio or video
codec. If the endpoint is able to determine the list of available
codecs for circuit-switched media streams, it MUST use the
corresponding payload type numbers in the <fmt> subfield.

In some cases, the endpoint is not able to determine the list of
available codecs for circuit-switched media streams. In this case,
in order to be syntactically compliant with SDP [RFC4566], the
endpoint MUST include a single dash ("-") in the <fmt> subfield.

As per RFC 4566 [RFC4566], the media format descriptions are listed
in priority order.

Examples of media descriptions for circuit-switched audio streams
are:

    m=audio 9 PSTN 3 0 8

    m=audio 9 PSTN -

Similarly, an example of a media description for circuit-switched
video stream is:

    m=video 9 PSTN 34

    m=video 9 PSTN -

5.2.3. Correlating the PSTN Circuit-Switched Bearer with SDP

   The endpoints should be able to correlate the circuit-switched bearer
   with the session negotiated with SDP in order to avoid ringing for an
   incoming circuit-switched bearer that is related to the session
   controlled with SDP (and SIP).

   Several alternatives exist for performing this correlation. This
   memo provides three mutually non-exclusive correlation mechanisms.
   Additionally, we define a fourth mechanism where correlation may be
   performed by external means, typically by the human user, in case
   using other correlation mechanisms is not possible or does not
   succeed. Other correlation mechanisms may exist, and their usage
   will be specified when need arises.

All mechanisms share the same principle: some unique information is
sent in the SDP and in the circuit-switched signaling protocol. If
these pieces of information match, then the circuit-switched bearer
is part of the session described in the SDP exchange. Otherwise,
there is no guarantee that the circuit-switched bearer is related to
such session.

The first mechanism is based on the exchange of PSTN Caller ID
between the endpoints. The Caller ID is also available as the
Calling Party Number in the circuit-switched signaling.

The second mechanism is based on the inclusion in SDP of a value that
is also sent in the User-User Information Element that is part of the
bearer setup signaling in the PSTN.

The third mechanism is based on sending in SDP a string that
represents Dual-Tone Multi-Frequency (DTMF) digits that will be later
sent right after the circuit-switched bearer is established.

The fourth correlation mechanism declares support for cases where
correlation is done by external means. Typically, this means that
the decision is left to the human user. This is how some current
conferencing systems operate: after logging on to the conference, the
system calls back to the user's phone number to establish audio
communications, and it is up to the human user to accept or reject
the incoming call. By declaring explicit support for this mechanism,
endpoints can use it only when such a possibility exists.

Endpoints may opt to implement any combination of the correlation
mechanisms specified in Sections 5.2.3.2, 5.2.3.3, 5.2.3.4, and
5.2.3.5, including the option to implement none at all.

5.2.3.1. The "cs-correlation" Attribute

In order to provide support for the correlation mechanisms, we define
a new media-level SDP attribute called "cs-correlation". There MUST
be at most one "cs-correlation" attribute per media description.

This "cs-correlation" attribute MAY contain zero or more subfields --
"callerid", "uuie", "dtmf", or "external" to specify additional
information required by the Caller ID, User-User Information Element,
DTMF, or external correlation mechanisms, respectively. The list of
correlation mechanisms may be extended by other specifications; see
Section 5.2.3.6 for more details.

The following sections provide more detailed information about these
subfields.

The values "callerid", "uuie", "dtmf", and "external" refer to the
correlation mechanisms defined in Sections 5.2.3.2, 5.2.3.3, 5.2.3.4,
and 5.2.3.5, respectively. The formal Augmented Backus-Naur Format
(ABNF) syntax of the "cs-correlation" attribute is presented in
Section 5.7.

5.2.3.2. Caller ID Correlation Mechanism

The Caller ID correlation mechanism consists of an exchange of the
Calling Party Number as an international E.164 number in SDP,
followed by the availability of the Calling Party Number Information
Element in the call setup signaling of the circuit-switched
connection. If both pieces of information match, the circuit-
switched bearer is correlated to the session described in SDP.

An example of inclusion of an international E.164 number in the
"cs-correlation" attribute is:

    a=cs-correlation:callerid:+441134960123

The presence of the "callerid" subfield indicates that the endpoint
supports use of the Calling Party Number as a means of correlating a
PSTN call with the session being negotiated. The "callerid" subfield
MAY be accompanied by the international E.164 number of the party
inserting the parameter.

   Note that there are no guarantees that this correlation mechanism
   works or is even available, due a number of problems:

   * The endpoint might not be aware of its own E.164 number, in
      which case it cannot populate the SDP appropriately.

   * The Calling Party Number Information Element in the circuit-
      switched signaling might not be available, e.g., due to policy
      restrictions of the network operator or caller restriction due
      to privacy.

   * The Calling Party Number Information Element in the circuit-
      switched signaling might be available, but the digit
      representation of the E.164 number might differ from the one
      expressed in the SDP, due to, e.g., lack of country code. To
      mitigate this problem, implementations should consider only
      some of the rightmost digits from the E.164 number for
      correlation. For example, the numbers +44-113-496-0123 and
      0113-496-0123 could be considered as the same number. This is
      also the behavior of some cellular phones, which correlate the
      incoming calling party with a number stored in the phone book,

        for the purpose of displaying the caller's name. Please refer
        to ITU-T E.164 recommendation [ITU.E164.2010] for consideration
        of the relevant number of digits to consider.

5.2.3.3. User-User Information Element Correlation Mechanism

   A second correlation mechanism is based on including in SDP a string
   that represents the User-User Information Element that is part of the
   call setup signaling of the circuit-switched bearer. The User-User
   Information Element is specified in ITU-T Q.931 [ITU.Q931.1998] and
   3GPP TS 24.008 [TS.24.008], among others. The User-User Information
   Element has a maximum size of 35 or 131 octets, depending on the
   actual message of the PSTN protocol where it is included and the
   network settings.

   The mechanism works as follows. An endpoint creates a User-User
   Information Element, according to the requirements of the call setup
   signaling protocol. The same value is included in the SDP offer or
   SDP answer, in the "uuie" subfield of the "cs-correlation" attribute.
   When the SDP offer/answer exchange is completed, each endpoint has
   become aware of the value that will be used in the User-User
   Information Element of the call setup message of the PSTN protocol.
   The endpoint that initiates the call setup attempt includes this
   value in the User-User Information Element. The recipient of the
   call setup attempt can extract the User-User Information Element and
   correlate it with the value previously received in the SDP. If both
   values match, then the call setup attempt corresponds to that
   indicated in the SDP.

   According to ITU-T Q.931 [ITU.Q931.1998], the User-User Information
   Element (UUIE) identifier is composed of a first octet identifying
   this as a User-User Information Element, a second octet containing
   the length of the user-user contents, a third octet containing a
   Protocol Discriminator, and a value of up to 32 or 128 octets
   (depending on network settings) containing the actual User
   Information (see Figure 4-36 in [ITU.Q931.1998]). The first two
   octets of the UUIE MUST NOT be used for correlation; only the octets
   carrying the Protocol Discriminator and the User Information value
   are input to the creation of the value of the "uuie" subfield in the
   "cs-correlation" attribute. Therefore, the value of the "uuie"
   subfield in the "cs-correlation" attribute MUST start with the
   Protocol Discriminator octet, followed by the User Information
   octets. The value of the Protocol Discriminator octet is not
   specified in this document; it is expected that organizations using
   this technology will allocate a suitable value for the Protocol
   Discriminator.

Once the binary value of the "uuie" subfield in the "cs-correlation"
attribute is created, it MUST be base 16 (also known as "hex")
encoded before it is inserted in SDP. Please refer to RFC 4648
[RFC4648] for a detailed description of base 16 encoding. The
resulting encoded value needs to have an even number of hexadecimal
digits and MUST be considered invalid if it has an odd number.

> Note: The encoding of the "uuie" subfield of the "cs-correlation"
> attribute is largely inspired by the encoding of the same value in
> the User-to-User header field in SIP, according to "A Mechanism
> for Transporting User to User Call Control Information in SIP"
> [SIP-UUI].

As an example, an endpoint willing to send a UUIE containing a
Protocol Discriminator with the hexadecimal value of %x56 and an
hexadecimal User Information value of %xA390F3D2B7310023 would
include an "a=cs-correlation" attribute line as follows:

    a=cs-correlation:uuie:56A390F3D2B7310023

Note that the value of the User-User Information Element is
considered as an opaque string and only used for correlation
purposes. Typically, call signaling protocols impose requirements on
the creation of a User-User Information Element for end-user protocol
exchange. The details regarding the generation of the User-User
Information Element are outside the scope of this specification.

Please note that there are no guarantees that this correlation
mechanism works. On one side, policy restrictions might not make the
User-User information available end to end in the PSTN. On the other
hand, the generation of the User-User Information Element is
controlled by the PSTN circuit-switched call protocol, which might
not offer enough freedom for generating different values from one
endpoint to another one or from one call to another in the same
endpoint. This might result in the same value of the User-User
Information Element for all calls.

5.2.3.4. DTMF Correlation Mechanism

We introduce a third mechanism for correlating the circuit-switched
bearer with the session described with SDP. This is based on
agreeing on a sequence of digits that are negotiated in the SDP
offer/answer exchange and sent as DTMF tones as described in ITU-T
Recommendation Q.23 [ITU.Q23.1988] over the circuit-switched bearer
once this bearer is established. If the DTMF digit sequence received
through the circuit-switched bearer matches the digit string
negotiated in the SDP, the circuit-switched bearer is correlated with

the session described in the SDP. The mechanism is similar to many
voice conferencing systems that require the user to enter a PIN code
using DTMF tones in order to be accepted in a voice conference.

The mechanism works as follows. An endpoint selects a DTMF digit
sequence. The same sequence is included in the SDP offer or SDP
answer, in a "dtmf" subfield of the "cs-correlation" attribute. When
the SDP offer/answer exchange is completed, each endpoint has become
aware of the DTMF sequence that will be sent right after the circuit-
switched bearer is set up. The endpoint that initiates the call
setup attempt sends the DTMF digits according to the procedures
defined for the circuit-switched bearer technology used. The
recipient (passive side of the bearer setup) of the call setup
attempt collects the digits and compares them with the value
previously received in the SDP. If the digits match, then the call
setup attempt corresponds to that indicated in the SDP.

   Note: Implementations are advised to select a number of DTMF
   digits that provide enough assurance that the call is related but
   do not prolong the bearer setup time unnecessarily. A number of 5
   to 10 digits is a good compromise.

As an example, an endpoint willing to send DTMF tone sequence "14D*3"
would include an "a=cs-correlation" attribute line as follows:

   a=cs-correlation:dtmf:14D*3

If the endpoints successfully agree on the usage of the DTMF digit
correlation mechanism but the passive side does not receive any DTMF
digits after successful circuit-switched bearer setup or receives a
set of DTMF digits that do not match the value of the "dtmf"
attribute (including receiving too many digits), the passive side
SHOULD consider that this DTMF mechanism has failed to correlate the
incoming call.

5.2.3.5. External Correlation Mechanism

The fourth correlation mechanism relies on external means for
correlating the incoming call to the session. Since endpoints can
select which correlation mechanisms they support, it may happen that
no other common correlation mechanism is found or that the selected
correlation mechanism does not succeed due to the required feature
not being supported by the underlying PSTN network. In these cases,
the human user can make the decision to accept or reject the incoming
call, thus "correlating" the call with the session. Since not all
endpoints are operated by a human user and since there may be no

other external means implemented by the endpoint for the correlation
function, we explicitly define support for such an external
correlation mechanism.

Endpoints wishing to use this external correlation mechanism would
use the "external" subfield in the "cs-correlation" attribute.
Unlike the other three correlation mechanisms, the "external"
subfield does not accept a value. The following is an example of an
"a=cs-correlation" attribute line:

    a=cs-correlation:external

Endpoints that are willing to only use the three explicit correlation
mechanisms defined in this document ("callerid", "uuie", and/or
"dtmf") would not include the "external" mechanism in the
offer/answer exchange.

The external correlation mechanism typically relies on the human user
to make the decision on whether or not the call is related to the
ongoing session. After the user accepts the call, that bearer is
considered as related to the session. There is a small chance that
the user receives at the same time another circuit-switched call that
is not related to the ongoing session. The user may reject this call
if he is able to determine (e.g., based on the calling line
identification) that the call is not related to the session and
continue waiting for another call attempt. If the user accepts the
incoming circuit-switched call, but it turns out to be not related to
the session, the endpoints need to rely on the human user to take
appropriate action (typically, the user would hang up).

5.2.3.6. Extensions to Correlation Mechanisms

New values for the "cs-correlation" attribute may be specified. The
registration policy for new values is "Specification Required"; see
Section 8. Any such specification MUST include a description of how
the SDP offer/answer mechanism is used to negotiate the use of the
new values, taking into account how endpoints determine which side
will become active or passive (see Section 5.3 for more details).

If, during the offer/answer negotiation, either endpoint encounters
an unknown value in the "cs-correlation" attribute, it MUST consider
that mechanism as unsupported and MUST NOT include that value in
subsequent offer/answer negotiation.

5.3. Negotiating the Correlation Mechanisms

   The four correlation mechanisms presented above (based on Called
   Party Number, User-User Information Element, DTMF digit sending, and
   external) are non-exclusive and can be used independently of each
   other. In order to know how to populate the "cs-correlation"
   attribute, the endpoints need to agree which endpoint will become the
   active party, i.e., the one that will set up the circuit-switched
   bearer.

5.3.1. Determining the Direction of the Circuit-Switched Bearer Setup

   In order to avoid a situation where both endpoints attempt to
   initiate a connection simultaneously, the direction in which the
   circuit-switched bearer is set up MUST be negotiated during the
   offer/answer exchange.

   The framework defined in RFC 4145 [RFC4145] allows the endpoints to
   agree which endpoint acts as the active endpoint when initiating a
   TCP connection. While RFC 4145 [RFC4145] was originally designed for
   establishing TCP connections, it can be easily extrapolated to the
   connection establishment of circuit-switched bearers. This
   specification uses the concepts specified in RFC 4145 [RFC4145] for
   agreeing on the direction of establishment of a circuit-switched
   bearer.

   RFC 4145 [RFC4145] defines two new attributes in SDP: "setup" and
   "connection". The "setup" attribute indicates which of the endpoints
   should initiate the connection establishment of the PSTN circuit-
   switched bearer. Four values are defined in Section 4 of RFC 4145
   [RFC4145]: "active", "passive", "actpass", and "holdconn". Please
   refer to Section 4 of RFC 4145 [RFC4145] for a detailed description
   of this attribute.

   The "connection" attribute indicates whether a new connection is
   needed or an existing connection is reused. The attribute can take
   the values "new" or "existing". Please refer to Section 5 of RFC
   4145 [RFC4145] for a detailed description of this attribute.

   Implementations that are compliant with this specification MUST
   support the "setup" and "connection" attributes specified in RFC 4145
   [RFC4145], but applied to circuit-switched bearers in the PSTN.

   We define the active party as the one that initiates the circuit-
   switched bearer after the offer/answer exchange. The passive party
   is the one receiving the circuit-switched bearer. Either party may
   indicate its desire to become the active or passive party during the
   offer/answer exchange using the procedures described in Section 5.6.

5.3.2. Populating the "cs-correlation" Attribute

   By defining values for the subfields in the "cs-correlation"
   attribute, the endpoint indicates that it is willing to become the
   active party and that it can use those values in the Calling Party
   Number, in the User-User Information Element, or as DTMF tones during
   the circuit-switched bearer setup.

   Thus, the following rules apply:

   o An endpoint that can only become the active party in the circuit-
      switched bearer setup MUST include all correlation mechanisms it
      supports in the "cs-correlation" attribute and MUST also specify
      values for the "callerid", "uuie", and "dtmf" subfields. Notice
      that the "external" subfield does not accept a value.

   o An endpoint that can only become the passive party in the circuit-
      switched bearer setup MUST include all correlation mechanisms it
      supports in the "cs-correlation" attribute but MUST NOT specify
      values for the subfields.

   o An endpoint that is willing to become either the active or passive
      party (by including the "a=setup:actpass" attribute in the offer)
      MUST include all correlation mechanisms it supports in the
      "cs-correlation" attribute and MUST also specify values for the
      "callerid", "uuie", and "dtmf" subfields. Notice that the
      "external" subfield does not accept a value.

5.3.3. Considerations for Correlations

   Passive endpoints should expect an incoming circuit-switched (CS)
   call for setting up the audio bearer. Passive endpoints MAY suppress
   the incoming CS alert during certain time periods. Additional
   restrictions can be applied, such as the passive endpoint not
   alerting incoming calls originated from the number that was observed
   during the offer/answer negotiation.

   There may be cases when an endpoint is not willing to include one or
   more correlation mechanisms in the "a=cs-correlation" attribute line
   even if it supports it. For example, some correlation mechanisms can
   be omitted if the endpoint is certain that the PSTN network does not
   support carrying the correlation identifier. Also, since using the
   DTMF-based correlation mechanism requires the call to be accepted
   before DTMF tones can be sent, some endpoints may enforce a policy
   restricting this due to, for example, cost associated with received
   calls, making the DTMF-based mechanism unusable.

Note that it cannot be guaranteed that the correlation mechanisms
relying on caller identification, User-User Information Element, and
DTMF sending will succeed even if the usage of those was agreed
beforehand. This is due to the fact that correlation mechanisms
require support from the circuit-switched bearer technology used.

Therefore, even a single positive indication using any of these
mechanisms SHOULD be interpreted by the passive endpoint so that the
circuit-switched bearer establishment is related to the ongoing
session, even if the other correlation mechanisms fail.

If, after successfully negotiating any of the "callerid", "uuie", or
"dtmf" correlation mechanisms in the SDP offer/answer exchange, an
endpoint receives an incoming establishment of a circuit-switched
bearer with no correlation information present, the endpoint first
checks whether or not the offer/answer exchange was also used to
successfully negotiate the "external" correlation mechanism. If it
was, the endpoint should let the decision be made by external means,
typically the human user. If the "external" correlation mechanism
was not successfully negotiated, the endpoint should treat the call
as unrelated to the ongoing session in the IP domain.

5.4. Considerations for Usage of Existing SDP

5.4.1. Originator of the Session

According to SDP [RFC4566], the origin line in SDP has the following
syntax:

    o=<username> <sess-id> <sess-version> <nettype> <addrtype>
    <unicast-address>

Of interest here are the <nettype> and <addrtype> fields, which
indicate the type of network and type of address, respectively.
Typically, this field carries the IP address of the originator of the
session. Even if the SDP was used to negotiate an audio or video
media stream transported over a circuit-switched bearer, the
originator is using SDP over an IP bearer. Therefore, <nettype> and
<addrtype> fields in the "o=" line should be populated with the IP
address identifying the source of the signaling.

5.4.2. Contact Information

   SDP [RFC4566] defines the "p=" line, which may include the phone
   number of the person responsible for the conference. Even though
   this line can carry a phone number, it is not suited for the purpose
   of defining a connection address for the media. Therefore, we have
   selected to define the PSTN-specific connection addresses in the "c="
   line.

5.5. Considerations for Usage of Third Party Call Control (3PCC)

   "Best Current Practices for Third Party Call Control (3PCC) in the
   Session Initiation Protocol (SIP)" [RFC3725] outlines several flows
   that are possible in third party call control scenarios and
   recommends some flows for specific situations.

   One of the assumptions in [RFC3725] is that an SDP offer may include
   a "black hole" connection address, which has the property that
   packets sent to it will never leave the host that sent them. For
   IPv4, this "black hole" connection address is 0.0.0.0 or a domain
   name within the .invalid DNS top level domain.

   When using an E.164 address scheme in the context of third party call
   control, when the User Agent needs to indicate an unknown phone
   number, it MUST populate the <addrtype> of the SDP "c=" line with a
   "-" string.

      Note: This may result in the recipient of the initial offer
      rejecting such offer if the recipient of the offer was not aware
      of its own E.164 number. Consequently, it will not be possible to
      establish a circuit-switched bearer, since neither party is aware
      of its E.164 number.

5.6. Offer/Answer Mode Extensions

   In this section, we define extensions to the offer/answer model
   defined in "An Offer/Answer Model with the Session Description
   Protocol (SDP)" [RFC3264] to allow for PSTN addresses to be used with
   the offer/answer model.

5.6.1. Generating the Initial Offer

   The offerer, wishing to use PSTN audio or video stream, MUST populate
   the "c=" and "m=" lines as follows.

   The endpoint MUST set the <nettype> in the "c=" line to "PSTN" and
   the <addrtype> to "E164". Furthermore, the endpoint SHOULD set the
   <connection-address> field to its own international E.164 number
   (with a leading "+"). If the endpoint is not aware of its own E.164
   number, it MUST set the <connection-address> to "-".

   In the "m=" line, the endpoint MUST set the <media> subfield to
   "audio" or "video", depending on the media type, and the <proto>
   subfield to "PSTN". The <port> subfield SHOULD be set to "9" (the
   discard port). The values "audio" or "video" in the <media> subfield
   MUST NOT be set by the endpoint unless it has knowledge that these
   bearer types are available on the circuit-switched network.

   The <fmt> subfield carries the payload type number(s) the endpoint is
   wishing to use. Payload type numbers in this case refer to the
   codecs that the endpoint wishes to use on the PSTN media stream. For
   example, if the endpoint wishes to use the GSM codec, it would add
   payload type number 3 in the list of codecs. The list of payload
   types MUST only contain those codecs the endpoint is able to use on
   the PSTN bearer. In case the endpoint is not aware of the codecs
   available for the circuit-switched media streams, it MUST include a
   dash ("-") in the <fmt> subfield.

   The mapping table of static payload types numbers to payload types is
   initially specified in [RFC3551] and maintained by IANA. For dynamic
   payload types, the endpoint MUST define the set of valid encoding
   names and related parameters using the "a=rtpmap" attribute line.
   See Section 6 of RFC 4566 [RFC4566] for details.

   When generating the offer, the offerer MUST include an
   "a=cs-correlation" attribute line in the SDP offer. The offerer MUST
   NOT include more than one "cs-correlation" attribute per media
   description. The "a=cs-correlation" line SHOULD contain an
   enumeration of all the correlation mechanisms supported by the
   offerer, in the format of subfields. See Section 5.3.3 for more
   information on usage of the correlation mechanisms.

   The current list of subfields include "callerid", "uuie", "dtmf", and
   "external", and they refer to the correlation mechanisms defined in
   Sections 5.2.3.2, 5.2.3.3, 5.2.3.4, and 5.2.3.5, respectively.

If the offerer supports any of the correlation mechanisms defined in
this memo and is willing to become the active party, the offerer MUST
add the "callerid", "uuie", "dtmf", and/or "external" subfields and
MUST specify values for them as follows:

o The international E.164 number as the value in the "callerid"
   subfield.

o The contents of the User-User Information Element as the value of
   the "uuie" subfield.

o The DTMF tone string as the value of the "dtmf" subfield.

o The endpoint MUST NOT specify any value for the "external"
   subfield.

If the offerer is only able to become the passive party in the
circuit-switched bearer setup, it MUST add at least one of the
possible correlation mechanisms but MUST NOT specify values for those
subfields.

For example, if the offerer is willing to use the User-User
Information Element and DTMF digit-sending mechanisms but can only
become the passive party, and is also able to let the human user
decide whether the correlation should be done or not, it includes the
following lines in the SDP:

    a=cs-correlation:uuie dtmf external

    a=setup:passive

If, on the other hand, the offerer is willing to use the User-User
Information Element and the DTMF correlation mechanisms and is able
to become the active or passive side, and is also able to let the
human user decide whether the correlation should be done or not, it
includes the following lines in the SDP:

    a=cs-correlation:uuie:56A390F3D2B7310023 dtmf:14D*3 external

    a=setup:actpass

The negotiation of the value of the "setup" attribute takes place as
defined in Section 4.1 of RFC 4145 [RFC4145].

The offerer states which role or roles it is willing to perform; the
answerer, taking the offerer's willingness into consideration,
chooses which roles both endpoints will actually perform during the
circuit-switched bearer setup.

By "active" endpoint, we refer to an endpoint that will establish the circuit-switched bearer; by "passive" endpoint, we refer to an endpoint that will receive a circuit-switched bearer.

If an offerer does not know its international E.164 number, it MUST set the "setup" attribute to the value "active". If the offerer knows its international E.164 number, it SHOULD set the value to either "actpass" or "passive".

Also "holdconn" is a permissible value in the "setup" attribute. It indicates that the connection should not be established for the time being.

The offerer uses the "connection" attribute to decide whether a new circuit-switched bearer is to be established or not. For the initial offer, the offerer MUST use value "new".

5.6.2. Generating the Answer

If the offer contained a circuit-switched audio or video stream, the answerer first determines whether it is able to accept and use such streams on the circuit-switched network. If the answerer does not support or is not willing to use circuit-switched media for the session, it MUST construct an answer where the port number for such media stream(s) is set to zero, according to Section 6 of [RFC3264]. If the answerer is willing to use circuit-switched media for the session, it MUST ignore the received port number (unless the port number is set to zero).

If the offer included a "-" as the payload type number, it indicates that the offerer is not willing or able to define any specific payload type. Most often, a "-" is expected to be used instead of the payload type when the endpoint is not aware of or not willing to define the codecs that will eventually be used on the circuit-switched bearer. The circuit-switched signaling protocols have their own means of negotiating or indicating the codecs; therefore, an answerer SHOULD accept such offers and SHOULD set the payload type to "-" in the answer.

If the answerer explicitly wants to specify a codec for the circuit-switched media, it MAY set the respective payload numbers in the <fmt> subfield in the answer. This behavior, however, is NOT RECOMMENDED.

When receiving the offer, the answerer MUST determine whether it becomes the active or passive party.

If the SDP in the offer indicates that the offerer is only able to
become the active party, the answerer needs to determine whether it
is able to become the passive party. If this is not possible, e.g.,
due to the answerer not knowing its international E.164 number, the
answerer MUST reject the circuit-switched media by setting the port
number to zero on the answer. If the answerer is aware of its
international E.164 number, it MUST include the "setup" attribute in
the answer and set it to value "passive" or "holdconn". The answerer
MUST also include its E.164 number in the "c=" line.

If the SDP in the offer indicates that the offerer is only able to
become the passive party, the answerer MUST verify that the offerer's
E.164 number is included in the "c=" line of the offer. If the
number is included, the answerer MUST include the "setup" attribute
in the answer and set it to value "active" or "holdconn". If the
number is not included, the recipient of the offer is not willing to
establish a connection the E.164 based on a priori knowledge of cost,
or other reasons, call establishment is not possible, and the
answerer MUST reject the circuit-switched media by setting the port
number to zero in the answer.

If the SDP in the offer indicates that the offerer is able to become
either the active or passive party, the answerer determines which
role it will take. If the offer includes an international E.164
number in the "c=" line, the answerer SHOULD become the active party.
If the answerer does not become the active party and if the answerer
is aware of its E.164 number, it MUST become the passive party. If
the answerer does not become the active or the passive party, it MUST
reject the circuit-switched media by setting the port number to zero
in the answer.

For each media description where the offer includes a
"cs-correlation" attribute, the answerer MUST select from the offer
those correlation mechanisms it supports and include in the answer
one "a=cs-correlation" attribute line containing those mechanisms it
is willing to use. The answerer MUST only add one "cs-correlation"
attribute in those media descriptions where also the offer included a
"cs-correlation" attribute. The answerer MUST NOT add any mechanisms
that were not included in the offer. If there is more than one
"cs-correlation" attribute per media description in the offer, the
answerer MUST discard all but the first for any media description.
Also, the answerer MUST discard all unknown "cs-correlation"
attribute values.

If the answerer becomes the active party, it MUST add a value to any
of the possible subfields.

If the answerer becomes the passive party, it MUST NOT add any values to the subfields in the "cs-correlation" attribute.

After generating and sending the answer, if the answerer became the active party, it

o MUST extract the E.164 number from the "c=" line of the offer and MUST establish a circuit-switched bearer to that address.

o if the SDP answer contained a value for the "callerid" subfield, MUST set the Calling Party Number Information Element to that number.

o if the SDP answer contained a value for the "uuie" subfield, MUST send the User-User Information Element according to the rules defined for the circuit-switched technology used and set the value of the Information Element to that received in the SDP offer.

o if the SDP answer contained a value for the "dtmf" subfield, MUST send those DTMF digits according to the circuit-switched technology used.

If, on the other hand, the answerer became the passive party, it

o MUST be prepared to receive a circuit-switched bearer,

o if the offer contained a value for the "callerid" subfield, MUST compare that value to the Calling Party Number Information Element of the circuit-switched bearer. If the received Calling Party Number Information Element matches the value of the "callerid" subfield, the call SHOULD be treated as correlated to the ongoing session.

o if the offer contained a value for the "dtmf" subfield, MUST be prepared to receive and collect DTMF digits once the circuit-switched bearer is set up. The answerer MUST compare the received DTMF digits to the value of the "dtmf" subfield. If the received DTMF digits match the value of the "dtmf" subfield in the "cs-correlation" attribute, the call SHOULD be treated as correlated to the ongoing session.

o if the offer contained a value for the "uuie" subfield, MUST be prepared to receive a User-User Information Element once the circuit-switched bearer is set up. The answerer MUST compare the received UUIE to the value of the "uuie" subfield. If the value of the received UUIE matches the value of the "uuie" subfield, the call SHOULD be treated as correlated to the ongoing session.

   o if the offer contained an "external" subfield, MUST be prepared to
      receive a circuit-switched call and use the external means
      (typically, the human user) for accepting or rejecting the call.

   If the answerer becomes the active party, generates an SDP answer,
   and then it finds out that the circuit-switched call cannot be
   established, then the answerer MUST create a new SDP offer where the
   circuit-switched stream is removed from the session (actually, by
   setting the corresponding port in the "m=" line to zero) and send it
   to its counterpart. This is to synchronize both parties (and
   potential intermediaries) on the state of the session.

5.6.3. Offerer Processing the Answer

   When receiving the answer, if the SDP does not contain an
   "a=cs-correlation" attribute line, the offerer should take that as an
   indication that the other party does not support or is not willing to
   use the procedures defined in the document for this session and MUST
   revert to normal processing of SDP.

   When receiving the answer, the offerer MUST first determine whether
   it becomes the active or passive party, as described in
   Section 5.3.1.

   If the offerer becomes the active party, it

   o MUST extract the E.164 number from the "c=" line and MUST
      establish a circuit-switched bearer to that address.

   o if the SDP answer contained a value for the "uuie" subfield, MUST
      send the User-User Information Element according to the rules
      defined for the circuit-switched technology used and set the value
      of the Information Element to that received in the SDP answer.

   o if the SDP answer contained a value for the "dtmf" subfield, MUST
      send those DTMF digits according to the circuit-switched
      technology used.

   If the offerer becomes the passive party:

   o It MUST be prepared to receive a circuit-switched bearer.

   o Note that if delivery of the answer is delayed for some reason,
      the circuit-switched call attempt may arrive at the offerer before
      the answer has been processed. In this case, since the
      correlation mechanisms are negotiated as part of the offer/answer
      exchange, the answerer cannot know whether or not the incoming

circuit-switched call attempt is correlated with the session being
negotiated; thus, the offerer SHOULD answer the circuit-switched
call attempt only after it has received and processed the answer.

o If the answer contained a value for the "dtmf" subfield, the
   offerer MUST be prepared to receive and collect DTMF digits once
   the circuit-switched bearer is set up. The offerer SHOULD compare
   the received DTMF digits to the value of the "dtmf" subfield. If
   the received DTMF digits match the value of the "dtmf" subfield in
   the "cs-correlation" attribute, the call SHOULD be treated as
   correlated to the ongoing session.

o If the answer contained a value for the "uuie" subfield, the
   offerer MUST be prepared to receive a User-User Information
   Element once the circuit-switched bearer is set up. The offerer
   SHOULD compare the received UUIE to the value of the "uuie"
   subfield. If the value of the received UUIE matches the value of
   the "uuie" subfield, the call SHOULD be treated as correlated to
   the ongoing session.

o If the answer contained an "external" subfield, the offerer MUST
   be prepared to receive a circuit-switched call and use the
   external means (typically, the human user) for accepting or
   rejecting the call.

According the "An Offer/Answer Model with the Session Description
Protocol (SDP)" [RFC3264], the offerer needs to be ready to receive
media as soon as the offer has been sent. It may happen that the
answerer, if it became the active party, will initiate a circuit-
switched bearer setup that will arrive at the offerer before the
answer has arrived. However, the offerer needs to receive the answer
and examine the information about the correlation mechanisms in order
to successfully perform correlation of the circuit-switched call to
the session. Therefore, if the offerer receives an incoming circuit-
switched call, it MUST NOT accept the call before the answer has been
received. If no answer is received during an implementation-specific
time, the offerer MUST either modify the session according to
[RFC3264] or terminate it according to the session signaling
procedures in question (for terminating a SIP session, see Section 15
of [RFC3261]).

5.6.4. Modifying the Session

If, at a later time, one of the parties wishes to modify the session,
e.g., by adding a new media stream or by changing properties used on
an existing stream, it may do so via the mechanisms defined in "An
Offer/Answer Model with the Session Description Protocol (SDP)"
[RFC3264].

   If there is an existing circuit-switched bearer between the endpoints
   and the offerer wants to reuse that, the offerer MUST set the value
   of the "connection" attribute to "existing".

   If either party removes the circuit-switched media from the session
   (by setting the port number to zero), it MUST terminate the circuit-
   switched bearer using whatever mechanism is appropriate for the
   technology in question.

   If either party wishes to drop and reestablish an existing call, that
   party MUST first remove the circuit-switched media from the session
   by setting the port number to zero and then use another offer/answer
   exchange where it MUST set the "connection" attribute to "new". If
   the media types are different (for example, a different codec will be
   used for the circuit-switched bearer), the media descriptions for
   terminating the existing bearer and the new bearer can be in the same
   offer.

   If either party would like to remove existing RTP-based media from
   the session and replace that with a circuit-switched bearer, it would
   create a new offer to add the circuit-switched media as described in
   Section 5.6.1 above, replacing the RTP-based media description with
   the circuit-switched media description, as specified in RFC 3264
   [RFC3264].

   Once the offer/answer exchange is done, but the circuit-switched
   bearer is not yet established, there may be a period of time when no
   media is available. Also, it may happen that correlating the
   circuit-switched call fails for reasons discussed in Section 5.3.3.
   In this case, even if the offer/answer exchange was successful,
   endpoints are not able to receive or send media. It is up to the
   implementation to decide the behavior in this case; if nothing else
   is done, the user most likely hangs up after a while if there is no
   other media in the session. Note that this may also happen when
   switching from one RTP media to another RTP media (for example, when
   firewall blocks the new media stream).

   If either party would like to remove existing circuit-switched media
   from the session and replace that with RTP-based media, it would
   modify the media description as per the procedures defined in RFC
   3264 [RFC3264]. The endpoint MUST then terminate the circuit-
   switched bearer using whatever mechanism is appropriate for the
   technology in question.

5.7. Formal Syntax

   The following is the formal Augmented Backus-Naur Form (ABNF)
   [RFC5234] syntax that supports the extensions defined in this

   specification. The syntax is built above the SDP [RFC4566] and the
   tel URI [RFC3966] grammars. Implementations that are compliant with
   this specification MUST be compliant with this syntax.

   Figure 2 shows the formal syntax of the extensions defined in this
   memo.

```
        ; extension to the connection field originally specified
        ; in RFC 4566

        connection-field  = [%x63 "=" nettype SP addrtype SP
        connection-address CRLF]
        ; CRLF defined in RFC 5234

        ;nettype and addrtype are defined in RFC 4566

        connection-address =/  global-number-digits / "-"
        ; global-number-digits specified in RFC 3966

        ;subrules for correlation attribute
        attribute          =/ cs-correlation-attr
        ; attribute defined in RFC 4566
        cs-correlation-attr = "cs-correlation:" corr-mechanisms
        corr-mechanisms    = corr-mech *(SP corr-mech)
        corr-mech          = caller-id-mech / uuie-mech /
                              dtmf-mech / external-mech /
                              ext-mech
        caller-id-mech     = "callerid" [":" caller-id-value]
        caller-id-value    = "+" 1*15DIGIT
        ; DIGIT defined in RFC 5234
        uuie-mech          = "uuie" [":" uuie-value]
        uuie-value         = 1*65(HEXDIG HEXDIG)
                              ;This represents up to 130 HEXDIG
                              ; (65 octets)
                              ;HEXDIG defined in RFC 5234
                              ;HEXDIG defined as 0-9, A-F
        dtmf-mech          = "dtmf" [":" dtmf-value]
        dtmf-value         = 1*32(DIGIT / %x41-44 / %x23 / %x2A )
                              ;0-9, A-D, '#' and '*'
        external-mech      = "external"
        ext-mech           = ext-mech-name [":" ext-mech-value]
        ext-mech-name      = token
        ext-mech-value     = token
        ; token is specified in RFC 4566
```

                 Figure 2: Syntax of the SDP Extensions

6. Examples

   In the examples below, where an SDP line is too long to be displayed
   as a single line, a breaking character "\" indicates continuation in
   the following line. Note that this character is included for display
   purposes only. Implementations MUST write a single line without
   breaks.

6.1. Single PSTN Audio Stream

```
            Endpoint A                         Endpoint B
               |                                   |
               | (1) SDP offer (PSTN audio)        |
               |---------------------------------->|
               |                                   |
               | (2) SDP answer (PSTN audio)       |
               |<----------------------------------|
               |                                   |
               |     PSTN call setup               |
               |<----------------------------------|
               |                                   |
               |<==== media over PSTN bearer ====>|
               |                                   |
```
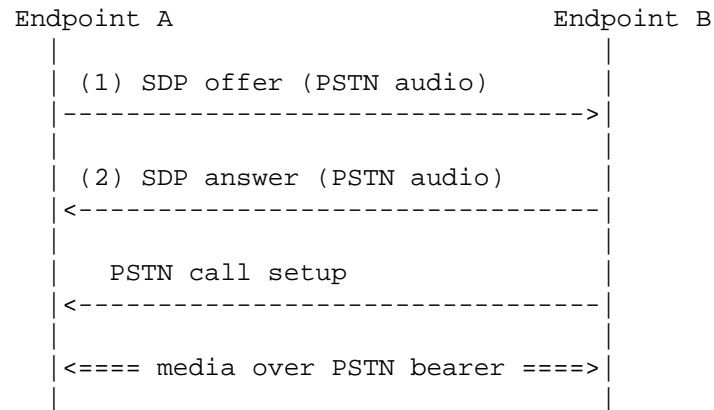
                         Figure 3: Basic Flow

   Figure 3 shows a basic example that describes a single audio media
   stream over a circuit-switched bearer. Endpoint A generates an SDP
   offer, which is shown in Figure 4. The offer describes a PSTN
   circuit-switched bearer in the "m=" and "c=" line where it also
   indicates its international E.164 number format. Additionally,
   Endpoint A expresses that it can initiate the circuit-switched bearer
   or be the recipient of it in the "a=setup" attribute line. The SDP
   offer also includes correlation identifiers that this endpoint will
   insert in the Calling Party Number and/or User-User Information
   Element of the PSTN call setup if eventually this endpoint initiates
   the PSTN call. Endpoint A also includes "external" as one
   correlation mechanism, indicating that it can use the human user to
   perform correlation in case other mechanisms fail.

```
v=0
o=alice 2890844526 2890842807 IN IP4 192.0.2.5
s=
t=0 0
m=audio 9 PSTN -
c=PSTN E164 +441134960123
a=setup:actpass
a=connection:new
a=cs-correlation:callerid:+441134960123 \
  uuie:56A390F3D2B7310023 external
```

                    Figure 4: SDP Offer (1)

   Endpoint B generates an SDP answer (Figure 5), describing a PSTN
   audio media on port 9 without information on the media subtype on the
   "m=" line. The "c=" line contains B's international E.164 number.
   In the "a=setup" line, Endpoint B indicates that it is willing to
   become the active endpoint when establishing the PSTN call, and it
   also includes the "a=cs-correlation" attribute line containing the
   values it is going to include in the Calling Party Number and User-
   User Information Element of the PSTN call establishment. Endpoint B
   is also able to perform correlation by external means, in case other
   correlation mechanisms fail.

```
v=0
o=- 2890973824 2890987289 IN IP4 192.0.2.7
s=
t=0 0
m=audio 9 PSTN -
c=PSTN E164 +441134960124
a=setup:active
a=connection:new
a=cs-correlation:callerid:+441134960124 \
  uuie:74B9027A869D7966A2 external
```

            Figure 5: SDP Answer with Circuit-Switched Media

   When Endpoint A receives the answer, it examines that B is willing to
   become the active endpoint when setting up the PSTN call. Endpoint A
   temporarily stores B's E.164 number and the User-User IE value of the
   "cs-correlation" attribute and waits for a circuit-switched bearer
   establishment.

   Endpoint B initiates a circuit-switched bearer using whatever
   circuit-switched technology is available for it. The Called Party
   Number is set to A's number, and the Calling Party Number is set to
   B's own number. Endpoint B also sets the User-User Information
   Element value to the one contained in the SDP answer.

When Endpoint A receives the circuit-switched bearer establishment,
it examines the UUIE and the Calling Party Number and, by comparing
those received during the offer/answer exchange, determines that the
call is related to the SDP session.

It may also be that neither the UUIE nor the Calling Party Number is
received by the called party, or the format of the Calling Party
Number is changed by the PSTN. Implementations may still accept such
call establishment attempts as being related to the session that was
established in the IP network. As it cannot be guaranteed that the
values used for correlation are always passed intact through the
network, they should be treated as additional hints that the circuit-
switched bearer is actually related to the session.

6.2. Advanced SDP Example: Circuit-Switched Audio and Video Streams

```
 Endpoint A                                    Endpoint B
    |                                              |
    | (1) SDP offer (PSTN audio and video)         |
    |--------------------------------------------->|
    |                                              |
    | (2) SDP answer (PSTN audio)                  |
    |<---------------------------------------------|
    |                                              |
    |    PSTN call setup                           |
    |<---------------------------------------------|
    |                                              |
    |<======== media over PSTN bearer ==========>|
    |                                              |
```
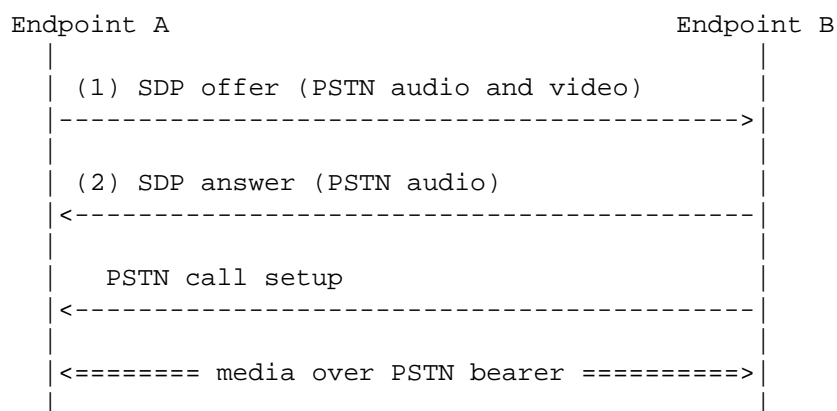
                Figure 6: Circuit-Switched Audio and Video Streams

Figure 6 shows an example of negotiating audio and video media
streams over circuit-switched bearers.

```
v=0
o=alice 2890844526 2890842807 IN IP4 192.0.2.5
s=
t=0 0
a=setup:actpass
a=connection:new
c=PSTN E164 +441134960123
m=audio 9 PSTN -
a=cs-correlation:dtmf:1234536
m=video 9 PSTN 34
a=rtpmap:34 H263/90000
a=cs-correlation:callerid:+441134960123
```

        Figure 7: SDP Offer with Circuit-Switched Audio and Video (1)

   Upon receiving the SDP offer described in Figure 7, Endpoint B
   rejects the video stream as the device does not currently support
   video, but it accepts the circuit-switched audio stream. As Endpoint
   A indicated that it is able to become either the active or passive
   party, Endpoint B gets to select which role it would like to take.
   Since the offer contained the international E.164 number of Endpoint
   A, Endpoint B decides that it becomes the active party in setting up
   the circuit-switched bearer. B includes a new value in the "dtmf"
   subfield of the "cs-correlation" attribute, which it is going to send
   as DTMF tones once the bearer setup is complete. The answer is
   described in Figure 8.

```
v=0
o=- 2890973824 2890987289 IN IP4 192.0.2.7
s=
t=0 0
a=setup:active
a=connection:new
c=PSTN E164 +441134960124
m=audio 9 PSTN -
a=cs-correlation:dtmf:654321
m=video 0 PSTN 34
a=cs-correlation:callerid:+441134960124
```

       Figure 8: SDP Answer with Circuit-Switched Audio and Video (2)

7. Security Considerations

   This document provides an extension to RFC 4566 [RFC4566] and RFC
   3264 [RFC3264]. As such, the security considerations of those
   documents apply.

This memo provides mechanisms to agree on a correlation identifier or
identifiers that are used to evaluate whether an incoming circuit-
switched bearer is related to an ongoing session in the IP domain.
If an attacker replicates the correlation identifier and establishes
a call within the time window the receiving endpoint is expecting a
call, the attacker may be able to hijack the circuit-switched bearer.
These types of attacks are not specific to the mechanisms presented
in this memo. For example, Caller ID spoofing is a well-known attack
in the PSTN. Users are advised to use the same caution before
revealing sensitive information as they would on any other phone
call. Furthermore, users are advised that mechanisms that may be in
use in the IP domain for securing the media, like Secure RTP (SRTP)
[RFC3711], are not available in the CS domain.

For the purposes of establishing a circuit-switched bearer, the
active endpoint needs to know the passive endpoint's phone number.
Phone numbers are sensitive information, and some people may choose
not to reveal their phone numbers when calling using supplementary
services like Calling Line Identification Restriction (CLIR) in GSM.
Implementations should take the caller's preferences regarding
calling line identification into account if possible, by restricting
the inclusion of the phone number in the SDP "c=" line if the caller
has chosen to use CLIR. If this is not possible, implementations may
present a prompt informing the user that their phone number may be
transmitted to the other party.

As with IP addresses, if there is a desire to protect the SDP
containing phone numbers carried in SIP, implementers are advised to
follow the security mechanisms defined in [RFC3261].

It is possible that an attacker creates a circuit-switched session
whereby the attacked endpoint should dial a circuit-switched number,
perhaps even a premium-rate telephone number. To mitigate the
consequences of this attack, endpoints MUST authenticate and trust
remote endpoints users who try to remain passive in the circuit-
switched connection establishment. It is RECOMMENDED that endpoints
have local policies precluding the active establishment of circuit-
switched connections to certain numbers (e.g., international,
premium, and long distance). Additionally, it is strongly
RECOMMENDED that the end user is asked for consent prior to the
endpoint initiating a circuit-switched connection.

8. IANA Considerations

   IANA has registered a number of SDP tokens according to the following
   data.

8.1. Registration of the New "cs-correlation" SDP Attribute

       Contact: Miguel Garcia <miguel.a.garcia@ericsson.com>

       Attribute name: cs-correlation

       Long-form attribute name: PSTN Correlation Identifier

       Type of attribute: media level only

       Subject to charset: No

       Description: This attribute provides the Correlation Identifier
       used in PSTN signaling

       Appropriate values: see Section 5.2.3.1

       Specification: RFC 7195

   The IANA has created a subregistry for the "cs-correlation" attribute
   under the "Session Description Protocol (SDP) Parameters" registry.
   The initial values for the subregistry are presented in the
   following; IANA has registered these values accordingly:

   Value of "cs-correlation" attribute Reference Description
   ----------------------------------- --------- -------------------
   callerid   RFC 7195 Caller ID
   uuie   RFC 7195 User-User
                                                 Information Element
   dtmf   RFC 7195 Dual-Tone
                                                 Multi-Frequency
   external   RFC 7195 External

   As per the terminology in [RFC5226], the registration policy for new
   values of the "cs-correlation" attribute is "Specification Required".

8.2. Registration of a New "nettype" Value

   IANA has registered a new "nettype" in the "Session Description
   Protocol (SDP) Parameters" registry [IANA]. The registration data,
   according to RFC 4566 [RFC4566], is as follows.

   Type   SDP Name   Reference
   --------------   ------------------   ---------
   nettype   PSTN   RFC 7195

8.3. Registration of a New "addrtype" Value

   IANA has registered a new "addrtype" in the "Session Description
   Protocol (SDP) Parameters" registry [IANA]. The registration data,
   according to RFC 4566 [RFC4566], is as follows.

   Type   SDP Name   Reference
   --------------   ------------------   ---------
   addrtype   E164   RFC 7195

   Note: This document defines the "E164" addrtype in the context of the
   "PSTN" nettype only. RFC 3108 [RFC3108] also defines address type
   "E.164". This definition is distinct from the one defined by this
   memo and shall not be used with <nettype> "PSTN".

8.4. Registration of a New "proto" Value

   IANA has registered a new "proto" in the "Session Description
   Protocol (SDP) Parameters" registry [IANA]. The registration data,
   according to RFC 4566 [RFC4566], is as follows.

   Type   SDP Name   Reference
   --------------   ------------------   ---------
   proto   PSTN   RFC 7195

   The related "fmt" namespace reuses the conventions and payload type
   number defined for RTP/AVP. In this document, the RTP audio and
   video media types, when applied to PSTN circuit-switched bearers,
   represent merely an audio or video codec in its native format
   directly on top of a single PSTN bearer.

   In some cases, the endpoint is not able to determine the list of
   available codecs for circuit-switched media streams. In this case,
   in order to be syntactically compliant with SDP [RFC4566], the
   endpoint MUST include a single dash ("-") in the <fmt> subfield.

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