Session Description Protocol (SDP) Extension For Setting Up Audio Media
 Streams Over Circuit-Switched Bearers In The Public Switched Telephone
                              Network (PSTN)
                       draft-garcia-mmusic-sdp-cs-02

Status of this Memo

Abstract

   This memo describes use cases, requirements, and protocol extensions
   for using the Session Description Protocol (SDP) Offer/Answer model
   for establishing audio 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 other transport protocols 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 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 media, and using a circuit-switched bearer
would offer, however, 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 is received
through the classical circuit-switched bearer.

Setting up a signaling relationship in the IP domain instead of just
setting up a circuit-switched call offers also 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 media
streams over to the high-speed bearer. This implies a new exchange

of SDP offer/answer that lead to a re-negotiation of the media
streams.

Other use cases exists. For example, and endpoint might have at its
disposal circuit-switch and packet-switched connectivity, but the
audio codecs are not the same in both access networks. Consider that
the circuit-switched audio stream supports narrow-bandwidth codecs,
while the packet-switched access allows any other audio 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 a
few examples of descriptions of circuit-switched audio streams in
SDP. Section 7 and Section 8 contain the IANA and Security
considerations, respectively.

2. Conventions Used in This Document

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT",
"SHOULD", "SHOULD 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 media stream over circuit-switched bearers.

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

REQ-2: The mechanism MUST allow the endpoints to combine circuit-
        switched audio 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 connection, i.e., which
         endpoint is active when initiating the circuit-switched
         connection.

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 call with the one negotiated in
         SDP, as opposed to another incoming call that is not related
         to that.

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

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

4. Overview of Operation

The mechanism defined in this memo extends SDP and allows describing
an audio media stream established over a circuit-switched bearer.
New tokens are registered in the "c=" and "m=" lines to be able to
describe an audio media stream over a circuit-switched bearer. These
SDP extensions are described in Section 5.2. Since circuit-switched
bearers are a sort of connection-oriented media streams, the
mechanism re-uses 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 5.3.3.

4.1. Example Call Flow

Consider the example presented in Figure 1. In this example, Alice
is located in an environment where she has access to both IP and
circuit-switched bearers for communicating with other endpoints.
Alice 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 circuit-
switched bearer.

```
                  Alice                              Bob
                    | (1) SDP Offer (PSTN audio)       |
                    |--------------------------------->|
                    |                                  |
                    | (2) SDP Answer (PSTN audio)      |
                    |<---------------------------------|
                    |                                  |
                    |    PSTN call setup               |
                    |<---------------------------------|
                    |                                  |
                    |                                  |
                    |<===== media over PSTN bearer ====>|
                    |                                  |
```
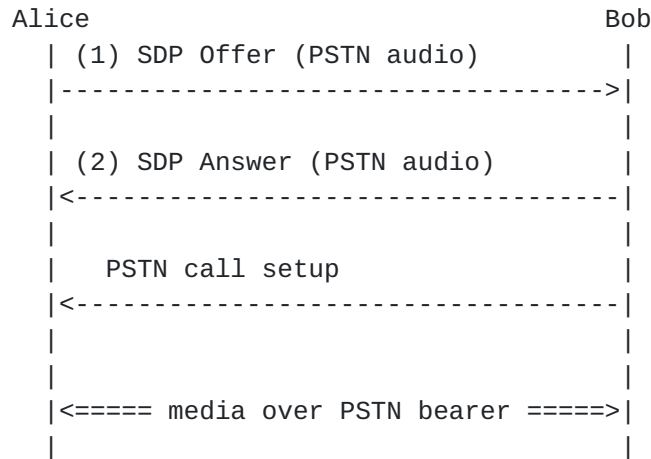
                    Figure 1: Example Flow

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

   During the offer-answer exchange Alice and Bob also agree the
   direction in which the circuit-switched connection should be
   established. The exchange also contains identifiers or references
   that can be used on the circuit-switched network for addressing the
   other endpoint, as well as identifying that the incoming circuit-
   switched bearer establishment is related to the ongoing session
   between Alice and Bob.

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

   Alice accepts the circuit-switched connection; the circuit-switched
   bearer setup is completed. Bob and Alice can now use the circuit-
   switched connection for two-way audio media.

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

5. Protocol Description

5.1. Level of Compliance

Implementations according to this specification MUST implement the
SDP extensions described in Section 5.2, and MUST implement the
considerations discussed in Section 5.3.

5.2. Extensions to SDP

This section provides the syntax and semantics of the extensions
required for providing a description of audio 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 the <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 consider the possibility of
describing E.164 telephone numbers. We define a new "E164" address
type. When used, the "E164" address type indicates that the
connection address contains a telephone number represented according
to the ITU-T E.164 [ITU.E164.1991] recommendation.

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

Note that <addrtype> and/or <connection-address> should not be
omitted without being set to a "-" since this would violate basic
syntax of SDP [RFC4566].

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

    c=PSTN E164 +15551234

    c=PSTN - -

5.2.2. Media Descriptions

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

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

The <media> sub-field carries the media type. Since this document
deals with establishing an audio bearer, the existing "audio" media
type is used.

The <port> sub-field is the transport port to which the media stream
is sent. Circuit-switched access lacks the concept of a port number,
and therefore the <port> sub-field is set to the discard port "9".

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

In the case of circuit-switched descriptions, RTP is not really used.
Rather than specifying the RTP audio profile payload type, we use the
<fmt> sub-field to indicate the list of available media types over

the circuit-switched bearer. Therefore, the <fmt> sub-field MAY
indicate one or more available audio codecs for a circuit-switched
audio stream. We use the classical RTP audio media types, even when
applied to PSTN circuit-switched bearers, the media type merely
represents an audio codec.

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

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

Example of a media description for circuit-switched audio streams is:

    m=audio 9 PSTN 3 0 8

    m=audio 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 to avoid ringing for an incoming
circuit-switched call that is related to the session controlled with
SDP (and SIP).

Several alternatives exist for performing this correlation. This
memo provides two mutually non-exclusive correlation mechanisms.
Other correlation mechanisms might exist as well, and their usage
will be specified when need arises. The first mechanism is based on
the exchange of PSTN caller-ID between the endpoints, which is also
made available as the Calling Party ID in the circuit-switched
signaling. The second mechanism is based on the inclusion in SDP of
the value of the User-to-User Information Element that is part of the
call setup signaling in the PSTN. Implementations MAY use any of
these mechanisms and MAY use both mechanisms simultaneously.

5.2.3.1. Caller-ID Correlation Mechanism

The caller-ID correlation mechanisms consists of an exchange of the
calling party number in E.164 format in SDP, followed by the
availability of the Calling Party Number information element in the
call setup signaling of the circuit switched connection.

An endpoint that is feasible to become the active party for setting
up the circuit-switched bearer SHOULD include its E.164 number in the
<connection-address> field of the "c=" line. An endpoint that acts

   as the passive party for setting up the circuit-switch bearer MAY
   include its E.164 number in the same corresponding lines, although
   these are not used for correlation.

   Example of inclusion of E.164 number in the "c=" line is:

      c=PSTN E164 +15551234

   Please note that there are no warranties that this correlation
   mechanism works or is even available, due a number of problems:

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

   o 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.

   o 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. For example, one can be represented in
      international format and the other might only contain the
      significant national digits. Therefore, an implementation may
      consider only some of the rightmost digits from the E.164 number
      for correlation. For example, the numbers +358-1-555-12345 and
      01-555-12345 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.

5.2.3.2. **User-User Information Element Correlation Mechanism**

   A second correlation mechanism is based on indicating in SDP 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 [3GPP.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.

   The mechanism works as follows: An endpoint creates a User-User
   Information Element, according to the requirement of the call setup
   signaling protocol. The same value is included in the SDP offer or
   SDP answer, in a new attribute called 'corr-id', defined below. 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 establishment 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 attempt corresponds to that indicated in the
SDP.

Note that, for correlation purposes, the value of the User-User
Information Element is considered as a opaque string and only used
for correlation purposes. Typically call signaling protocols impose
requirements on the creation of 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.

This specification defines a new SDP attribute, called 'corr-id',
whose purpose is to include the User-User Information Element that
the endpoint will include in the call setup attempt. The 'corr-id'
attribute has the following format:

    a=corr-id:corr-token

An endpoint that is feasible to become the active party for setting
up the PSTN call SHOULD include in the 'corr-id' attribute the value
of the User-User Information Element that will be used in the PSTN
call setup attempt. If both the SDP offerer and the SDP answerer are
able to become the active party, each one SHOULD include a
correlation value. Then the party that becomes active in setting up
the PSTN circuit-switched call includes this value in the User-User
information element of the call signaling setup. The passive party
is able to inspect the received value of User-User Information
Element and correlate it with that received in the SDP in the
'corr-id' attribute. An endpoint that takes the role of the passive
party for setting up the circuit-switched bearer MAY include a
'corr-id' attribute in the SDP, although it is not used for
correlation purposes.

Please note that there are no warranties 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.

5.3. Considerations for Usage of Existing SDP

5.3.1. 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 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.

5.3.2. Contact Information

SDP [RFC4566] defines the "p=" line which may include the phone
number of the person reponsible 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.

5.3.3. Determining the Direction for Setting Up the Circuit-Switched
         Bearer

Either endpoint can initiate the establishment of the circuit-
switched bearer. 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 should 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 is 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", "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 according to this specification MUST support the 'setup' and 'connection' attributes specified in RFC 4145 [RFC4145], but applied to circuit-switched bearers in the PSTN.

In order to establish a circuit-switched connection in the PSTN, the initiating endpoint needs to know the address (E.164 number) of the other endpoint. Therefore, if an endpoint wants to be able to receive incoming circuit-switched calls, it must know its E.164 number and must indicate it in SDP. As a consequence, an endpoint that is not aware of its own E.164 number cannot take the role of the passive side with respect the establishment of the circuit-switched connection.

5.4. 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] grammar. Implementations according to 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]
                         ;nettype and addrtype are defined in RFC 4566


   connection-address =  multicast-address / unicast-address /
                         e164-address / "-"
                         ; multicast-address and unicast-address are
                         ; defined in RFC 4566

   e164-address       =  ["+"] 1*15DIGIT
                         ; DIGIT is specified in RFC 5234



   ;subrules for corr-id attribute
   attribute          =  corr-id-attr
                              ; attribute defined in RFC 4566

   corr-id-attr       =  "corr-id:" corr-id-value
   corr-id-value      =  1*32(ALPHA/DIGIT)
```

                   Figure 2: Syntax of the SDP extensions

6. SDP Examples

6.1. Basic SDP Example: Single Circuit-Switched Audio Stream

```
               Alice                             Bob
                 |                                |
                 | (1) SDP Offer (PSTN audio)     |
                 |------------------------------->|
                 |                                |
                 | (2) SDP Answer (PSTN audio)    |
                 |<-------------------------------|
                 |                                |
                 |    PSTN call setup             |
                 |<-------------------------------|
                 |                                |
                 |<==== media over PSTN bearer ====>|
                 |                                |
```
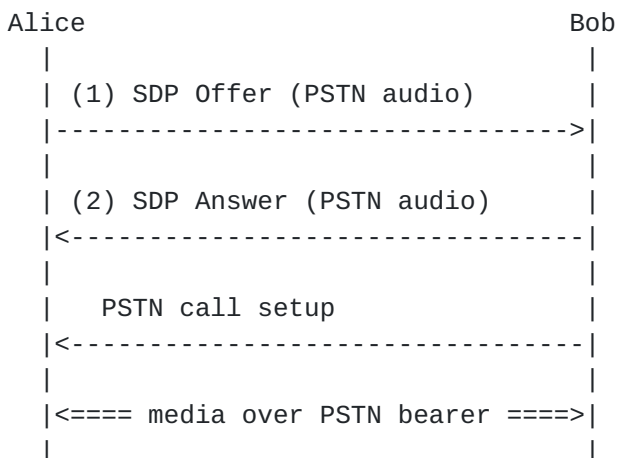
                      Figure 3: Basic flow

   Figure 3 shows a basic example that describes a single audio media
   stream over a circuit-switched bearer. The SDP offer is show in

Figure 4. The endpoint describes a PSTN circuit-switched bearer in
the "m=" and "c=" line where it also indicates its E.164 number.
Additionally, it expresses that it can initiate the circuit-switched
connection or be the recipient of it. The SDP offer also includes a
correlation identifier that this endpoint will be inserting the User-
User Information Element of the PSTN call setup if eventually this
endpoint initiates the PSTN call.

```
v=0
o=jdoe 2890844526 2890842807 IN IP4 192.0.2.5
s=
t=0 0
m=audio 9 PSTN -
c=PSTN E164 +15551234
a=setup:actpass
a=connection:new
a=corr-id:2890W284hAT452612908awudfjang908
```

                    Figure 4: SDP offer (1)

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

```
          Alice                                Bob
            |                                   |
            | (1) SDP Offer (IP and PSTN audio)|
            |--------------------------------->|
            |                                   |
            | (2) SDP Answer (PSTN audio)      |
            |<---------------------------------|
            |                                   |
            |    PSTN call setup               |
            |<---------------------------------|
            |                                   |
            |<==== media over PSTN bearer ====>|
            |                                   |
```
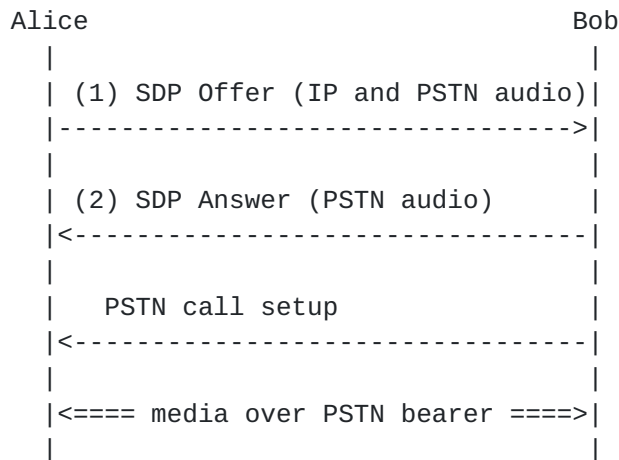
                  Figure 5: Alternative media

Figure 5 shows an example of negotiating audio media streams over IP
or circuit-switched bearers. Using the mechanisms described in SDP
Capability Negotiation Framework
[I-D.ietf-mmusic-sdp-capability-negotiation] and extensions thereof
(SDP media capabilities Negotiation
[I-D.ietf-mmusic-sdp-media-capabilities] and SDP Miscellaneous
Capabilities [I-D.garcia-mmusic-sdp-misc-cap]) it is possible to
construct an SDP offer where audio media can be offered alternatively
over IP or circuit-switched bearer.

```
v=0
o=jdoe 2890844526 2890842807 IN IP4 192.0.2.5
s=
t=0 0
m=audio 49170 RTP/AVP 0 8 3
c=IN IP4 192.0.2.5
a=creq:med-v0,ccap-v0
a=mcap:1 PCMU/8000/1
a=mcap:2 PCMA/8000/1
a=mcap:3 GSM/8000/1
a=mcap:4 -
a=tcap:1 RTP/AVP PSTN
a=ccap:1 IN IP4 192.0.2.1
a=ccap:2 PSTN E164 +15551234
a=acap:1 corr-id:2890W284hAT452612908awudfjang908
a=acap:2 setup:actpass
a=acap:3 connection:new
a=pcfg:1 t=1 m=1,2,3 c=1
a=pcfg:2 t=2 m=4 c=2 a=1,2,3
```

              Figure 6: SDP offer with alternative media (1)

Upon receiving the SDP offer descibed in Figure 6, Bob decided to
select the circuit-switched bearer and generates the answer described
in Figure 7

```
v=0
o=- 2890973824 2890987289 IN IP4 192.0.2.7
s=
t=0 0
m=audio - PSTN -
c=PSTN - -
a=acfg:2
a=setup:active
a=connection:new
a=corr-id:2890W284hAT452612908awudfjang908
```

           Figure 7: SDP answer with circuit-switched media (2)

7. IANA Considerations

   This document instructs IANA to register a number of SDP tokens
   according to the following data.

7.1. **Registration of a New 'corr-id' SDP Attribute**

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

       Attribute name: corr-id

       Long-form attribute name: PSTN Correlation Identifier

       Type of attribute: media level only

       This attribute is subject to the charset attribute

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

       Specification: RFC XXXX

7.2. **Registration of a New "nettype" value**

   This memo provides instructions to IANA to register a new "nettype"
   in the Session Description Protocol Parameters registry [1]. The
   registration data, according to RFC 4566 [RFC4566] follows.

   Type   SDP Name   Reference
   ----   ------------------   ---------
   nettype   PSTN   [RFCxxxx]

7.3. **Registration of New "addrtype" values**

   This memo provides instructions to IANA to register a new "addrtype"
   in the Session Description Protocol Parameters registry [1]. The
   registration data, according to RFC 4566 [RFC4566] follows.

   Type   SDP Name   Reference
   ----   ------------------   ---------
   addrtype   E164   [RFCxxxx]
                   -                           [RFCxxxx]

7.4. **Registration of a New "proto" value**

   This memo provides instructions to IANA to register a new "proto" in
   the Session Description Protocol Parameters registry [1]. The
   registration data, according to RFC 4566 [RFC4566] follows.

   Type   SDP Name   Reference
   -------------- -------------------------- ---------
   proto   PSTN   [RFCxxxx]

8. Security Considerations

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

9. Acknowledgments

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