Codec Capabilities Attribute for SDP

Status of this Memo

1. Abstract

The need for assured communication arisen protocols like RSVP [2]
and SBM [3] which are linked to access technologies to guarantee the
delay and drop probabilities of a given IP flow.

One of the side-effects of the assurance is that when the bandwidth
of the IP flow goes outside the contracted region then the IP flow
will be disturbed and the communication will be effected.

This document discusses the effects of assured communication to SDP
protocol, identifies the weaknesses and proposes a new attribute to
rectify these weaknesses.

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 RFC-2119 [4].

4. **Introduction**

   The introduction of commercial services (e.g. VoIP) brought along
   with the quality concept. Although the quality has many dimensions
   one of the basic aspects of quality is the packet delay and drop
   probabilities.

   Resource Reservation Protocol (RSVP) answers these basic QoS
   requirements by making a contract with the client. Using RSVP
   protocol the client defines its IP flow using a set of filters and
   defines its IP flow by using a set of measures.

   When the Router accepts the contract it is assumed that as far as
   the client is within the contracted parameters it will get the
   requested QoS.

   The Session Description Protocol [5] which is designed for the
   advertisement of conference sessions and communicate the relevant
   conference setup information to recipients. SDP by definition does
   not contain elements that would enable the end-points for media
   encoding negotiations.

   In this document a new attribute Codec Capabilities is defined such
   that end-points will know codec choices at the same time having
   definite codec(s) agreed for communication.

5. **The Current SDP Codec Communication**

   Several drafts are written to identify and solve problems involving
   QoS setup in IP telephony [6, 7, 8]. In all of the drafts that are
   cited it is assumed that the both ends of communication will have a
   solid understanding of which codec the other side will use.

   For example consider the call flow of Figure 1 which is taken from
   [6] which uses SIP messaging [9].

```
                  A                              B
                  ----------INV-(1)------------>
                  <------180 w/SDP-(2)-----------
                  <.....PATH..B2A.................
                  ..........RESV.B2A............>

                  ..........PATH.A2B............>

                  <,,,,,,,,RESVCONF B2A,,,,,,,,,,
        LOCAL RB  <........RESV...A2B............

                  ,,,,,,,,,,,,RESVCONF A2B,,,,,,>  RINGING

                  <-------------- 200 OK-(3)----
```

```
                 --------------ACK ------------>
```

For the case of simplicity lets assume that both A and B has two
codec's G.726 for voice communications and G.711 for fax and modem.
In this case the SIP messages INVITE, 183 and 200 OK will contain
the following SDP's.

INVITE (1) SDP:
```
        v=0
        o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
        s=TakeMeHome
        c=IN IP4 224.2.17.12/127
        t=907165245 0
        m=audio 49170 RTP/AVP 0 96
        a=rtpmap:0 PCMU/8000
        a=rtpmap:96 G726-32/8000
        a=qos:mandatory sendrecv
```

183 (2) SDP:
```
        v=0
        o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
        s=WannaGoHome
        c=IN IP4 224.2.17.12/127
        t=907165245 0
        m=audio 49170 RTP/AVP 0 96
        a=rtpmap:0 PCMU/8000
        a=rtpmap:96 G726-32/8000
        a=qos:mandatory sendrecv
```

200OK (3) SDP:
```
        v=0
        o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
        s=WannaGoHome
        c=IN IP4 224.2.17.12/127
        t=907165245 0
        m=audio 49170 RTP/AVP 0 96
        a=rtpmap:0 PCMU/8000
        a=rtpmap:96 G726-32/8000
        a=qos:mandatory sendrecv
```

At the end of the messaging both of the ends have information
regarding both ends, meaning G.711 and G.726 codecs.

The problem is that if both ends does not have any kind of pre-
agreement than although both ends will use 8Kb/s bandwidth they has
to reserve 64Kb/s since they cannot control when will the other end
switch to the high bandwidth codec. Which would mean that there will
be an over-reservation of 56Kb/s.

. **Codec Capabilities**

The _m=_ Media Announcements has double interpretations: first it contains the capabilities of the receiver, and second it contains the expected Codec(s) at the same time.

The proposed attribute codec Capabilities takes the double meaning from Media Announcements. Codec Capabilities shows to the other end the originators capabilities and preferences.

[7](#). **Definition**

The Codec Capabilities attribute is defined as:

    a= cap: <media> <port> <transport> <Codec Capabilities List>

The definitions of <media> <port> and <transport> are the same as Media Announcement and they are used for identifying the correct media announcement for the cases that there are more than one media announcement.

The <Codec Capabilities> is an preference list containing:

    <Codec Capabilities List>
                    = <Payload Type>["/"<Direction>["/"<Group>]]

The leftmost item in the list is the most desired codec to be used with the order of decreasing reference moving right in the list.

The <Payload Type> contains the payload types that the originator is willing to send and/or receive.

The <Direction> field is optional and if exists it MUST contain one of the following:

    r: if the client is willing to receive (default).
    s: if the client is willing to send.
    a: if the client is willing to send and/or receive.
    b: if the client is willing to both send and receive.

The nonexistence of <Direction> field means the client is willing to receive.

The difference between "a" and "b" is the fact that if "a" is used it is possible for a media exchange in one direction and with a different selection in the other direction. If "b" is to be used the media exchange can only take place if both directions use the same codec type.

The <Group> field is optional and if exists it is used for tying multiple media streams together. If there is no grouping for the payload type than either 0 SHOULD be used or the field MUST be

omitted.

[8](8). Examples

[8.1](8.1). IP Telephony

If the previous example is to be re-written:

INVITE SDP:
```
     v=0
     o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
     s=WannaGoHome
     c=IN IP4 224.2.17.12/127
     t=907165245 0
     m=audio 49170 RTP/AVP 0
     a=cap: audio 49170 RTP/AVP 0/a/0 96/a/0
     a=rtpmap:0 PCMU/8000
     a=rtpmap:96 G726-32/8000
     a=qos:mandatory sendrecv
```

183 SDP:
```
     v=0
     o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
     s=WannaGoHome
     c=IN IP4 224.2.17.12/127
     t=907165245 0
     m=audio 49170 RTP/AVP 0
     a=cap: audio 49170 RTP/AVP 0/a/0 96/a/0
     a=rtpmap:0 PCMU/8000
     a=rtpmap:96 G726-32/8000
     a=qos:mandatory sendrecv
```

200OK SDP:
```
     v=0
     o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
     s=WannaGoHome
     c=IN IP4 224.2.17.12/127
     t=907165245 0
     m=audio 49170 RTP/AVP 0
     a=cap: audio 49170 RTP/AVP 0/a/0 96/a/0
     a=rtpmap:0 PCMU/8000
     a=rtpmap:96 G726-32/8000
     a=qos:mandatory sendrecv
```

[8.2](8.2). UAS with Limited Processing Power

If the UAS has limited processing power and cannot handle both
sending and receiving of low bandwidth codecs simultaneously then
the SDP will be:

```
    INVITE SDP:
          v=0
          o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
          s=WannaGoHome
```
```
          c=IN IP4 224.2.17.12/127
          t=907165245 0
          m=audio 49170 RTP/AVP 0
          a=cap: audio 49170 RTP/AVP 0/b/0 96/s/1 0/r/1 0/s/2 96/r/2
          a=rtpmap:0 PCMU/8000
          a=rtpmap:96 G726-32/8000
          a=qos:mandatory sendrecv
```

8.3. Sending Capabilities without Tying the Originator Codec

In all of the examples above the originator set the Codec it wants
to receive without knowing the other ends capabilities. The only way
to achieve this is to have symmetric codec definition for bth send
and receive dircetions such that the receiving end chooses the codec
for both directions. The SDP exchange will be:

```
    INVITE SDP:
          v=0
          o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
          s=WannaGoHome
          c=IN IP4 224.2.17.12/127
          t=907165245 0
          m=audio 49170 RTP/AVP 0 96
          a=cap: audio 49170 RTP/AVP 0/b/0 96/b/0
          a=rtpmap:0 PCMU/8000
          a=rtpmap:96 G726-32/8000
          a=qos:mandatory sendrecv


    183 SDP:
          v=0
          o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
          s=WannaGoHome
          c=IN IP4 224.2.17.12/127
          t=907165245 0
          m=audio 49170 RTP/AVP 0
          a=cap: audio 49170 RTP/AVP 0/a/0 96/a/0
          a=rtpmap:0 PCMU/8000
          a=rtpmap:96 G726-32/8000
          a=qos:mandatory sendrecv
```

At the end of 183 exchange both ends has the codec(s) that will be
used both directions for media exchange, and originator has the
knowledge that the terminating end can support any codec
combination, and the preferred codec is G.726.

8.4. Codec change

The codec change by the consent of other end can be achieved by
changing the first payload type defined in the Codec Capablities
List. If the above example the originator decides that both ends
should change to a lower bandwidth codec G726 then the SDP exchange
will be:

    INVITE SDP:
          v=0
          o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
          s=WannaGoHome
          c=IN IP4 224.2.17.12/127
          t=907165245 0
          m=audio 49170 RTP/AVP 0
          a=cap: audio 49170 RTP/AVP 96/b/0 0/b/0
          a=rtpmap:0 PCMU/8000
          a=rtpmap:96 G726-32/8000
          a=qos:mandatory sendrecv

    183 SDP:
          v=0
          o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
          s=WannaGoHome
          c=IN IP4 224.2.17.12/127
          t=907165245 0
          m=audio 49170 RTP/AVP 96
          a=cap: audio 49170 RTP/AVP 0/a/0 96/a/0
          a=rtpmap:0 PCMU/8000
          a=rtpmap:96 G726-32/8000
          a=qos:mandatory sendrecv

Notice that the terminating end still shows its preference for the
higher bandwidth lower processing power codec keeping the preference
list the same.

8.5. Rejection of Codec Change

If in the above example the other end is to reject the request
because it does not have processing power than it would reply with
SDP:

    183 SDP:
          v=0
          o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
          s=WannaGoHome
          c=IN IP4 224.2.17.12/127
          t=907165245 0
          m=audio 49170 RTP/AVP 0

```
        a=cap: audio 49170 RTP/AVP 0/a/0 96/a/0
        a=rtpmap:0 PCMU/8000
        a=rtpmap:96 G726-32/8000
        a=qos:mandatory sendrecv
```

   In the above SDP the G.726 is still retained in the capabilities but
   it is possible for the other end not to use the codec anymore which
   would result with the SDP:

   183 SDP:

```
        v=0
        o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
        s=WannaGoHome
        c=IN IP4 224.2.17.12/127
        t=907165245 0
        m=audio 49170 RTP/AVP 0
        a=cap: audio 49170 RTP/AVP 0/a/0
        a=rtpmap:0 PCMU/8000
        a=rtpmap:96 G726-32/8000
        a=qos:mandatory sendrecv
```

8.6. Tying Multiple Media Types

   If there is a limited bandwidth between two end-points that is not
   sufficient for both high quality high bandwidth voice Codec and
   Video communications. Than the definition of SDP will be:

   High Quality Voice:
```
        v=0
        o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
        s=WannaGoHome
        c=IN IP4 224.2.17.12/127
        t=907165245 0
        m=audio 49170 RTP/AVP 99
        a=cap: audio 49170 RTP/AVP 99/a/0 0/a/1
        a=rtpmap:0 PCMU/8000
        a=rtpmap:99 VeryHighQualityCodec/64000
        a=qos:mandatory sendrecv
        a=cap: video 51372 RTP/AVP 31 b 1
```

   Both Voice and Video:
   INVITE:
```
        v=0
        o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
        s=WannaGoHome
        c=IN IP4 224.2.17.12/127
        t=907165245 0
        m=audio 49170 RTP/AVP 99
        a=cap: audio 49170 RTP/AVP 0/a/1 99/a/0
        a=rtpmap:0 PCMU/8000
```

```
        a=rtpmap:99 VeryHighQualityCodec/64000
        a=qos:mandatory sendrecv
        a=cap: video 51372 RTP/AVP 31 b 1


    183/200:
        v=0
        o=MotherShip 2987933615 2987933615 IN IP4 1.2.3.4
        s=TakeMeHome
        c=IN IP4 224.2.17.12/127
        t=907165245 0
        m=audio 49170 RTP/AVP 0
        a=cap: audio 49170 RTP/AVP 0/a/1 99/a/0
        a=rtpmap:0 PCMU/8000
        a=rtpmap:99 VeryHighQualityCodec/64000
```

```
        a=qos:mandatory sendrecv
        m= video 51372 RTP/AVP 31 b 1
        a=cap: video 51372 RTP/AVP 31 b 1
```

9. Security Considerations

If the contents of the SDP contained in the messages are private
then end-to-end encryption of the message body can be used to
prevent unauthorized access to the content.

10. References

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11.   Acknowledgments

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