Multiple Packetization Time in SDP Problem statement & Requirements

draft-garcia-mmusic-multiple-ptimes-problem-01.txt

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Problem

- SDP defines the ptime/maxptime
  - common parameter for all media formats in m-line
  - not possible to specify this in f(codec)

```
m=audio 49170 RTP/AVP 0 4 8
a=ptime:30
a=maxptime:60
```

    - codec 0, 8 (PCMU, PCMA): sample based codecs
      but some hardware have fixed 10 ms buffer
      RFC3551 defines a default 20 ms ptime
    - codec 4 (G723): speech frame of 30 ms
  - in case of IOT problems:
    mostly a quick, proprietary fix

Changes in version 01

- list of RFCs related to ptime/maxptime (definitions, recommendations, requirements, default value)
- Example of a problem case
- Requirements as indicated at IETF.69 meeting
  - limitations in DSP silicon
  - QoS budget calculations
  - Interworking issues
  - Codecs implemented in hardware
- A list of different solutions already proposed during the last years.

Different views

- ptime in f(codec)
  - Codec experts: no
  - IOT (SIPit): yes
  - SDO (Docsis, ITU): yes
  - SDP implementers: don’t know what to do
- Signalling
  - loose coupling - preference model (IETF model)
  - strong coupling - strict signaling (Telco model)

Requirements

- Codec experts
  - no requirements. ptime/maxptime is a system/network parameter related to packetization delay;
- IOT
  - Suffering from many different semantic interpretations.
- SDO
  - Provided own attributes to the SDP
    - ITU V.152: defines maxmptime attribute
      Maximum multiple ptime to indicate the supported packetization period for all codec payload types.
    - Docsis: defines mptime attribute
- SDP implementers
  - Asking for a “common” solution to avoid different semantics.

Some examples

- HW implementations issues
  - Many have fixed 20 ms buffer between codec/DAC.
  - Sample based codecs (G711): different buffer sizes in use (10 ms, 20 ms, dynamic)
    - problem case: G711, iLBC, ptime=20 but HW only supporting10 ms for G711
  - 20 ms is supported by most codecs.
    Frame based codecs of 30 ms frame size (iLBC, G723) gives problems.

- Docsis
  - Dynamic Timeslot

Cable networks (DOCSIS)

- Main requirement
  - Optimization of resource usage in a time-slot reservation system
- Cable modem <-> CMTS
  - CMTS = cable modem termination system
- Service grant system for upstream traffic
    - from CM to CMTS
    - CM request a “transmission timeslot” at CMTS
    - amount of bytes/timeslot reservation
- Main problem related to upstream
  - SDP capability negotiation?
  - for downstream: CMTS can schedule the transmission and take care of packet size changes

Real problem

- Semantic problem?
- Technical problem?
- Implementation issue?

- What will be solved by ptime=f(codec)?
  - requires negotiation instead of indication
  - min, max, range value
  - introduces more complexity/problems
  - what improvement can be obtained?

Next steps

- Further clarification with PacketCable
  - ptime, maxptime, bandwidth parameter
- Clarify if multiple ptimes is a standardization or implementation issue.
- Propose a solution for the use of ptime/maxptime in different scenario’s.