OsmoMSC - Bug #3650

MSC is not sending a payload_type to MNCC?

10/12/2018 01:35 PM - keith

Status:	Closed	**Start date:**	10/12/2018
Priority:	Normal	**Due date:**	
Assignee:	dexter	**% Done:**	0%
Category:			
Target version:			
Resolution:		**Spec Reference:**	

Description

The SDP created by osmo-sip-connector is invalid:

a=rtpmap:0 GSM/8000

Freeswitch rejects this.

Looks like the MSC is sending 0 for payload_type.
See mncc.c:check_rtp_create() in the sip cxtr

This works with legacy nitb:
You get this line in sip-connector's debug log:

mncc.c:393 RTP cnt leg(5010) ip(172.16.0.15), port(16412) pt(3) ptm(768)

With osmo-msc it will be:
mncc.c:393 RTP cnt leg(5010) ip(172.16.0.15), port(16412) pt(0) ptm(768)

note pt(0)

Related issues:

Related to osmo-sip-connector - Bug #3724: Wrong media format used in SIP INV...	**New**	**12/11/2018**
Related to osmo-sip-connector - Bug #1683: osmo-sip-connector: Implement code...	**Stalled**	**03/31/2016**

History

#1 - 10/12/2018 01:56 PM - laforge

- Assignee set to dexter

#2 - 11/04/2018 07:20 PM - neels

For 35c3 congress, this would be interesting to clarify...

#3 - 11/05/2018 04:59 PM - neels

let me copy the dirty hack mentioned on the ML here for later reference
(I haven't tested but wanted to find this if I need it.)

```
a quick and dirty hack for the osmo-sip-connector, to (probably)
get your calls running through FreeSwitch:

Hardcode override the pt in sdp_create_file() in sdp.c by adding

other->payload_type = 98; (or for full rate GSM, it would be
other->payload_type = 3;)

somewhere in the top of that function,

at line 170 for example, here:
http://git.osmocom.org/osmo-sip-connector/tree/src/sdp.c#n170)
```

#4 - 11/13/2018 11:11 AM - keith

neels wrote:

> For 35c3 congress, this would be interesting to clarify...

#3518 would also bite us at congress methinks..
and there's also a workaround:
https://gerrit.osmocom.org/#/c/osmo-sip-connector/+/11194/

#5 - 11/16/2018 03:37 PM - keith

Alternative:

In libmsc/gsm_04_08_cc.c (from line 1690), do:

```
    /* FIXME: This has to be set to some meaningful value,
     * before the MSC-Split, this value was pulled from
     * lchan->abis_ip.rtp_payload */
    uint32_t payload_type = 3;
```

#6 - 12/11/2018 05:20 PM - fixeria

- Related to Bug #3724: Wrong media format used in SIP INVITE causes one-way audio added

#7 - 10/04/2019 08:47 AM - keith

- Related to Bug #1683: osmo-sip-connector: Implement codec selection / move codec selection to osmo-msc added

#8 - 10/04/2019 08:49 AM - keith

- Status changed from New to Closed

As this is being resolved in #1683, I'll close this issue.