OsmoMSC - Bug #3735

call arriving during another call has no voice

12/16/2018 01:25 PM - neels

Status:	Stalled	**Start date:**	12/16/2018
Priority:	Normal	**Due date:**	
Assignee:	neels	**% Done:**	80%
Category:			
Target version:			
Resolution:			

Description

during testing for 35c3 POC reports:

- be on a voice call.
- second call comes in and knocks.
- hang up the first call and answer the second call.

Now the second call has no voice.

History

#1 - 12/16/2018 10:24 PM - neels

- Subject changed from call during call breaks first call, second call has no audio to call arriving during another call has no voice

- Description updated

#2 - 12/18/2018 01:13 AM - neels

- File second_call_no_voice_filtered.pcapng added

the conflict becomes apparent at msc_mgcp_call_assignment(), which is called for the new incoming call while the first call is still busy:
(see packet no. 149 in attached pcap)

```
        if (conn->rtp.mgcp_ctx) {
                LOGP(DMGCP, LOGL_ERROR, "(subscriber:%s) double assignment detected, dropping...\n",
                     vlr_subscr_name(trans->vsub));
                return -EINVAL;
        }
```

Per RAN connection, only one mgcp_ctx is available.
But if we are being called while another call is ongoing, there must be

- more than one mgcp_ctx,
  OR
- creating the mgcp_ctx must be postponed until the first call is terminated.

Later on, packet no. 228, the previous mgcp_ctx has been discarded and hence is NULL,
so completing the second call fails.

Goals:

- If the call assignment fails in this way, it should be dismantled and released. Currently it goes on silently.
- Successfully assign the second call.

#3 - 12/19/2018 01:30 PM - laforge

disclaimer: I don't know the current code off my head

On Tue, Dec 18, 2018 at 01:13:42AM +0000, neels [REDMINE] wrote:

> Per RAN connection, only one mgcp_ctx is available.
> But if we are being called while another call is ongoing, there must be
>
> - more than one mgcp_ctx,
>   OR
> - creating the mgcp_ctx must be postponed until the first call is terminated.

I doubt that either of the two resembles how GSM is specified. There can be up to 7
concurrent call control state machines on top of one MM/RR connection.

AFAIR, what we need is:

- context for the RAN/MS/UE side RTP connection (once)
- context for the CN side RTP connection (one for each call, up to 7 per subscriber)

when a new call comes in, we signal this to the user, and the user cann then use call-related
supplementary services to switch between the calls, for example

- put the first call on hold
- talk to the second call
- put the sceond call back on hold, talk to the first call, ...

I'm not saying we need this now, but this is what we should aim for if we make any changes
to data structures or architecture.

Until we support any of the above, the second call should probably simply be rejected
cleanly?

#4 - 12/20/2018 03:36 AM - neels

- Status changed from New to In Progress

With trivial code changes, I can manage to switch to the second call; that's on osmo-{bsc,msc} branches neels/call_during_call.
That's already a nice improvement, but uncovers an unclean teardown issue (if a transaction is gone during MGCP operation, the MGCP response tries to access the freed transaction).

Not working on master nor the patch branches is rejecting the second incoming call: the already ongoing call loses voice.

It's not urgent per se, but for example the POC at congress tries to call back after token activation, which falls on its face.

And also in general for everyone using osmo-msc and osmo-bsc, the user experience for a call during an ongoing call is currently dismal.
I would like to reach some non-dismal state of it for congress.

#5 - 12/21/2018 01:25 AM - neels

Ok! I've managed to fix both cases with fairly trivial code changes.

- The osmo-bsc must not attempt to re-use an existing lchan for Assignment Command, because re-routing RTP for that is currently not implemented.
  https://gerrit.osmocom.org/c/osmo-bsc/+/12401

- The MSC must not free MGCP/RTP-routing for an ongoing call when releasing a CC trans that has nothing to do with the ongoing call.
  https://gerrit.osmocom.org/c/osmo-msc/+/12397

- The MSC must postpone Assignment for a second incoming call until the first call has ended.
  https://gerrit.osmocom.org/c/osmo-msc/+/12400

After this:

- An incoming call can be rejected. The original call continues. (fixed)
- An incoming call can be taken while dropping the original call. (fixed)
- When taking an incoming call and putting the original call on "hold", a friendly message says that it is not possible,
  and the choice of receiving or rejecting the new call can still be made after that message, in a clean fashion.

I currently haven't tested yet what happens if there is **three** peers trying to talk to the same number, running out of lab phones (not really, just not bothering now).
Possibly we should be taking over the call indicated by the CC TIO, the current patch version takes on whichever waiting call it finds first.
This is fine if one call comes in on an ongoing call, and this is already a huge improvement to what we had before.

#6 - 12/21/2018 01:25 AM - neels

- % Done changed from 0 to 90

#7 - 12/21/2018 01:45 AM - neels

The patches currently fail to build because I'm already using LOGPFSMSL introduced in https://gerrit.osmocom.org/c/libosmocore/+/12386.
But they build and work fine.

#8 - 06/18/2019 09:18 PM - neels

- % Done changed from 90 to 80

this was resolved, but should be re-tested with new osmo-msc...
(Ideally with a ttcn3 test, but we don't verify the voice streaming in ttcn3 at all yet, so that would be rather difficult to introduce)

#9 - 06/18/2019 09:18 PM - neels

- Status changed from In Progress to Stalled

Files

second_call_no_voice_filtered.pcapng	46.4 KB	12/18/2018	neels