There is an initial implementation of GAPK based audio back-end in [fixeria/audio](https://github.com/fixeria/audio).

The current implementation is limited, so TODO/FIXME:

- init both I/O processing chains when CHANNEL MODE MODIFY is received,
- deinit both I/O chains when a call is finished,
- take care about PHY specific frame formats (e.g. TI),
- optimize the application loop in order to improve performance,
- support other than GSM FR codecs (currently FR is hardcoded),
- compose codec support in Classmark depending on PHY capabilities and GAPK codec support.

Some points require a possibility to know the PHY features, such as:

- TCH frame forwarding capability,
- preferred TCH frame format,
- full rate / half rate channel support,
- AMR codec over FR / HR channel support.
Revision 67d45c05 - 12/05/2022 11:28 PM - fixeria
mobile: do not enforce RTP format for Uplink TCH frames

This is a partial revert of 8f04fa9795819113f2a75a18303cbc3d688777f3d6.

The GAPK based audio I/O implementation of the mobile app is now capable of handling TI's specific TCH frame format, which can be configured via the VTY interface ('io-tch-format ti'). TCH frames in this format are different from the ones in RTP format and may have different length and different bit ordering. Remove voice_frame_verify().

Change-Id: I6113ba443e65d9ae091b643af54c873b7da4de8
Related: OS#3400

Revision 10eeb3ca - 12/05/2022 11:28 PM - fixeria
firmware: remove TCH/F specific bit re-ordering

This is a partial revert of d49a748cb8d777bb7f3d90617da8b462e6d1.

The GAPK based audio I/O implementation of the mobile app is now capable of handling TI's specific TCH frame format, which can be configured via the VTY interface ('io-tch-format ti'). Thus there is no need to have the conversion logic in the firmware anymore.

This patch also fixes the layer1 firmware, so it does not hang on receipt of Uplink TCH frames anymore when compiled with recent arm-none-eabi toolchain (v12.2.0 on my machine).

Change-Id: I5af6d4e4dd9c06d32768d01bc1e3e18d476706fb
Related: OS#3400

History

#1 - 07/16/2018 11:15 PM - fixeria
- Tracker changed from Bug to Feature

#2 - 07/16/2018 11:30 PM - fixeria
- Blocked by Feature #1461: include some version information / negotiation in the L1CTL protocol added

#3 - 08/27/2018 09:11 PM - fixeria
- Checklist item [x] Init both I/O processing chains when CHANNEL MODE MODIFY is received added
- Checklist item [x] Support other than GSM FR codecs (currently FR is hardcoded) added
- Checklist item [ ] Deinit both I/O chains when a call is finished added
- Checklist item [ ] Take care about PHY specific frame formats (e.g. TI) added
- Checklist item [ ] Optimize the application loop in order to improve performance added
- Checklist item [ ] Compose codec support in Classmark depending on PHY capabilities and GAPK codec support added

#4 - 08/27/2018 09:11 PM - fixeria
- Status changed from New to In Progress
- % Done changed from 0 to 50

#5 - 08/27/2018 09:16 PM - fixeria
- Checklist item [x] Optimize the application loop in order to improve performance set to Done

The performance problem is actually caused by blocking calls of src/pq_alsa.c/pq_cb_alsa_input()/snd_pcm_readi(). Setting both buffer and period size values seems to solve the issue. A separate issue needs to be created...

#6 - 10/04/2018 09:11 PM - fixeria
- Checklist item [ ] Investigate the problem with unpleasant audio effects added

#7 - 10/04/2018 09:11 PM - fixeria
- Status changed from In Progress to Stalled
After rebasing fixeria/audio on top of the recent master I got the audio working with Mot C1xx. Still need to test with trxcon. Will update commit messages and submit patches to Gerrit soon.

Commit messages updated, all patches submitted for review:

https://gerrit.osmocom.org/c/osmocom-bb/+/30325 mobile: add MNCC socket path to settings [NEW]
https://gerrit.osmocom.org/c/osmocom-bb/+/30326 mobile: add MNCC handler selection to settings [NEW]
https://gerrit.osmocom.org/c/osmocom-bb/+/30327 mobile: allow enabling/disabling handling of voice in the L1PHY [NEW]
https://gerrit.osmocom.org/c/osmocom-bb/+/30328 mobile: gsm48_rr_set_mode(): print name of channel mode [NEW]
https://gerrit.osmocom.org/c/osmocom-bb/+/30329 mobile: gsm48_rr_set_mode(): check rc of rsl_dec_chan_rr() [NEW]
https://gerrit.osmocom.org/c/osmocom-bb/+/30330 mobile: gsm48_rr_set_mode(): print error if ch_type is not TCH [NEW]
https://gerrit.osmocom.org/c/osmocom-bb/+/30331 mobile: gsm48_rr_set_mode(): fix copy-paste in comment [NEW]
https://gerrit.osmocom.org/c/osmocom-bb/+/30332 mobile: properly handle RR CHANNEL MODE MODIFY message [NEW]
https://gerrit.osmocom.org/c/osmocom-bb/+/30333 mobile: gsm48_rr_rx_voice(): drop TCH/F channel limitation [NEW]
https://gerrit.osmocom.org/c/osmocom-bb/+/30334 mobile: voice.h: use '#pragma once' include guard [NEW]
https://gerrit.osmocom.org/c/osmocom-bb/+/30335 mobile: voice.h: add missing forward declarations [NEW]
https://gerrit.osmocom.org/c/osmocom-bb/+/30336 mobile: split gsm_send_voice() -> gsm_send_voice_{msg,frame}() [NEW]
https://gerrit.osmocom.org/c/osmocom-bb/+/30337 mobile: integrate GAPK based audio (voice) I/O support [NEW]

I also force-pushed to https://cgit.osmocom.org/osmocom-bb/log/?h=fixeria/audio.

#12 - 11/26/2022 11:50 PM - fixeria
- Checklist item [ ] libosmo-gapk should be optional dependency added

#13 - 12/02/2022 11:59 AM - fixeria
- Checklist item [x] Get patches from @fixeria/audio@ merged to master set to Done

% Done changed from 50 to 60

All patches have been merged, so I removed https://cgit.osmocom.org/osmocom-bb/log/?h=fixeria/audio.

#14 - 12/05/2022 01:27 AM - fixeria
- Checklist item [x] Deinit both I/O chains when a call is finished set to Done

% Done changed from 60 to 80

https://gerrit.osmocom.org/c/osmocom-bb/+/30454 mobile: clean up GAPK I/O state on channel release [NEW]
https://gerrit.osmocom.org/c/osmocom-bb/+/30455 mobile: support RTP and Ti specific TCH frame I/O formats [NEW]

#15 - 12/05/2022 10:57 PM - fixeria
- Checklist item [x] AMR (AHS/AFS) support added

% Status changed from In Progress to Stalled

With a few additional patches applied:

https://gerrit.osmocom.org/c/osmocom-bb/+/30483 mobile: do not enforce RTP format for Uplink TCH frames [NEW]
https://gerrit.osmocom.org/c/osmocom-bb/+/30484 firmware: remove TCH/F specific bit re-ordering [NEW]
I have successfully tested GAPK based audio I/O with both trxcon and a Mot C1xx phone. The non-adaptive codecs (HR, FR, EFR) are all confirmed to work. AMR implementation is currently incomplete in trxcon, and is completely missing in the layer1 firmware (needs DSP patches); adding a checklist item for this. I think at the current state it qualifies as a working voice call functionality, so I am stopping to work on this ticket and setting it to Stalled.

IMHO it makes more sense to resolve this ticket once all patches for non-AMR are merged. Add a new issue for AMR support, with no assignee and low priority.

Ack, please see #5812.

All patches have been merged. Voice quality is good, except when using libgsmhr (via libosmo-gapk.so). This can be investigated later.