Design and implementation of an Audio codec (AMR-WB) using Data Flow Programming Language CAL in OpenDF environment.

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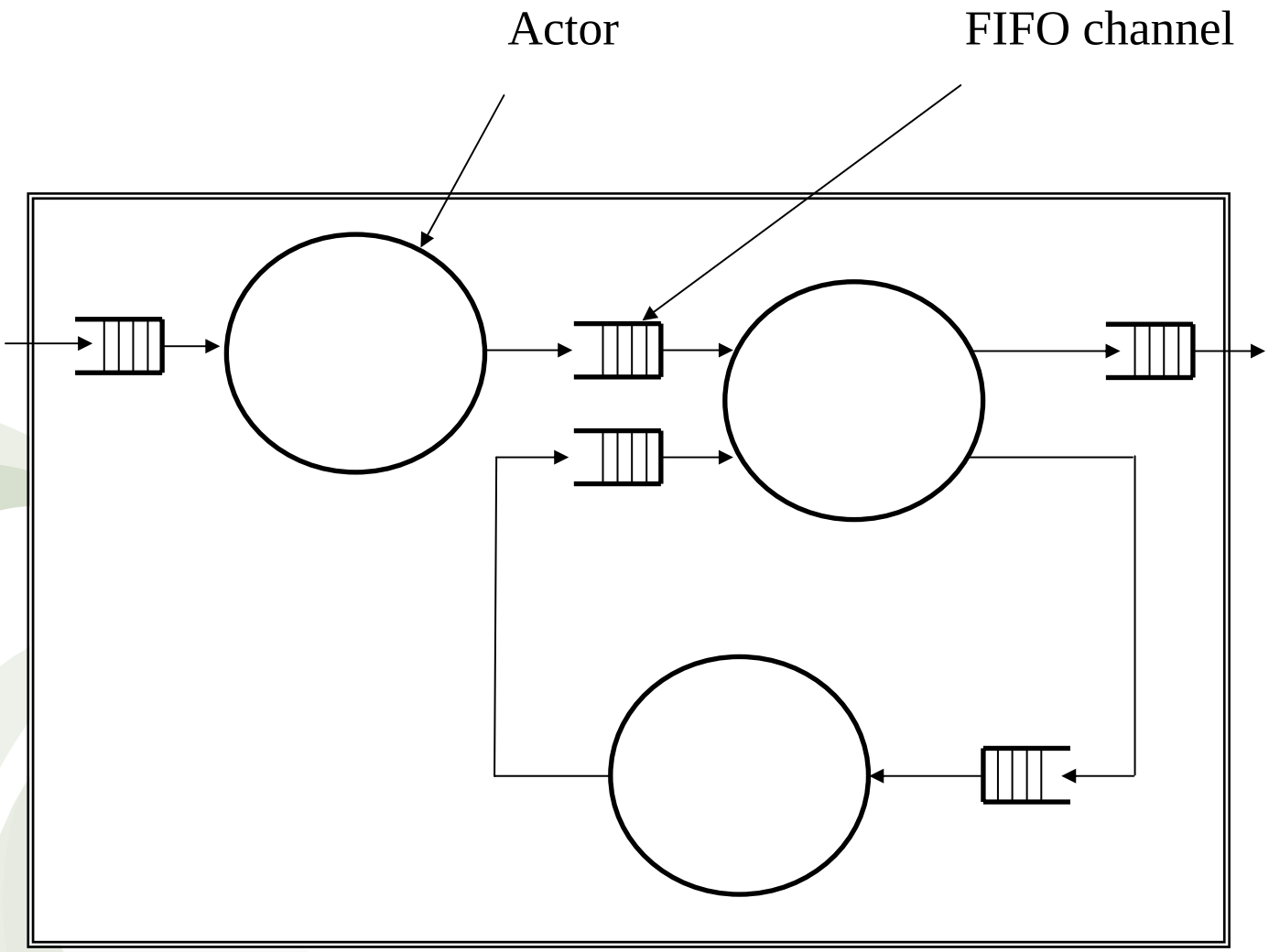
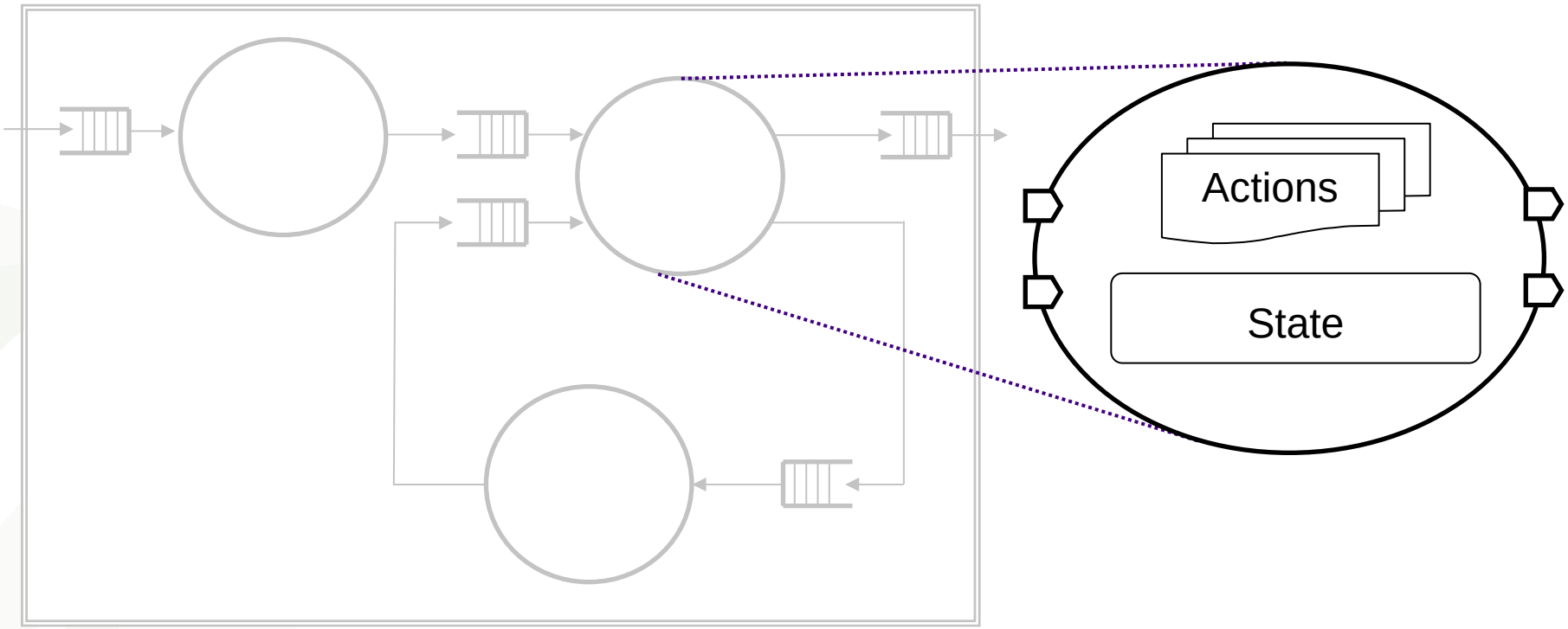
Dataflow

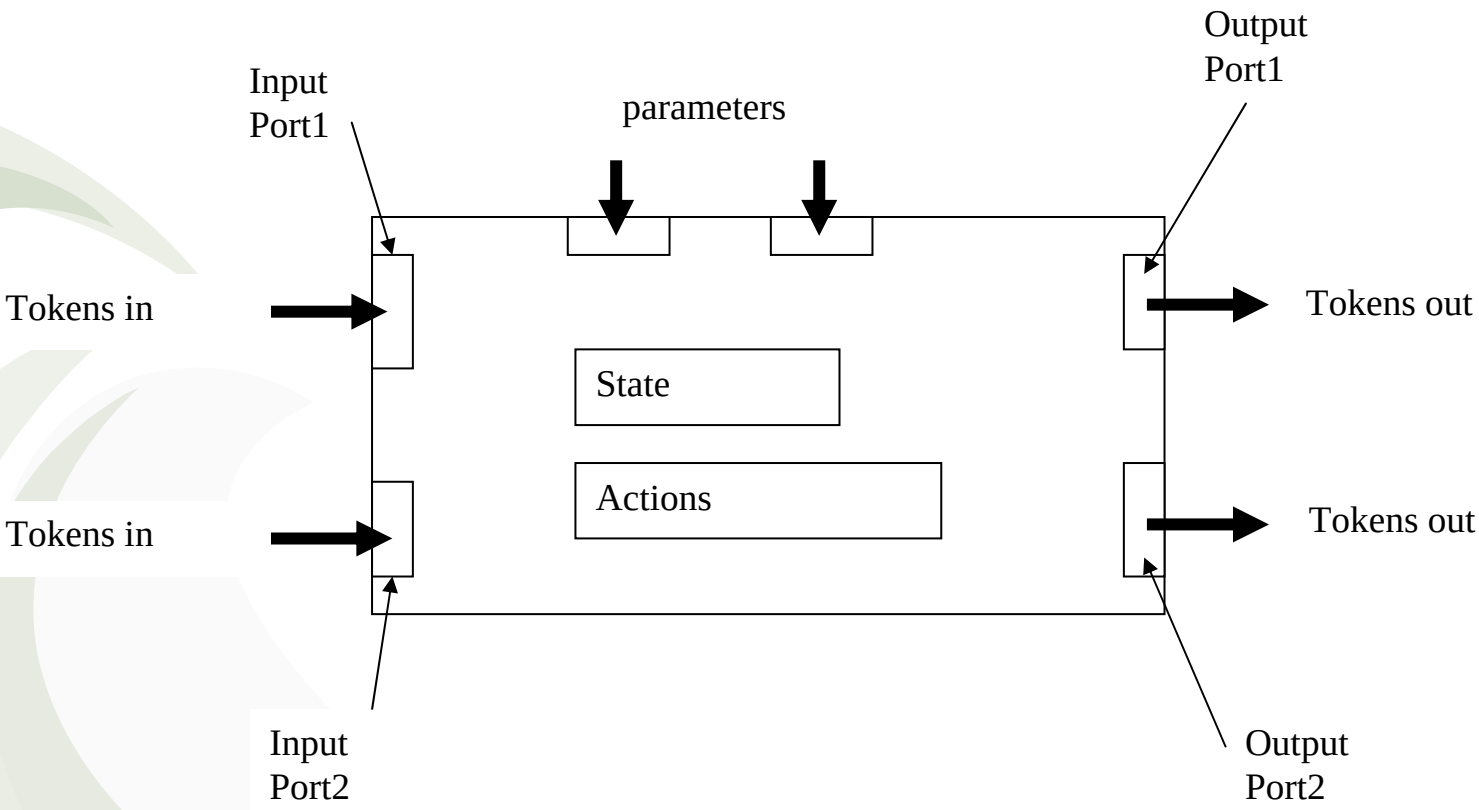


Actors & actions



CAL

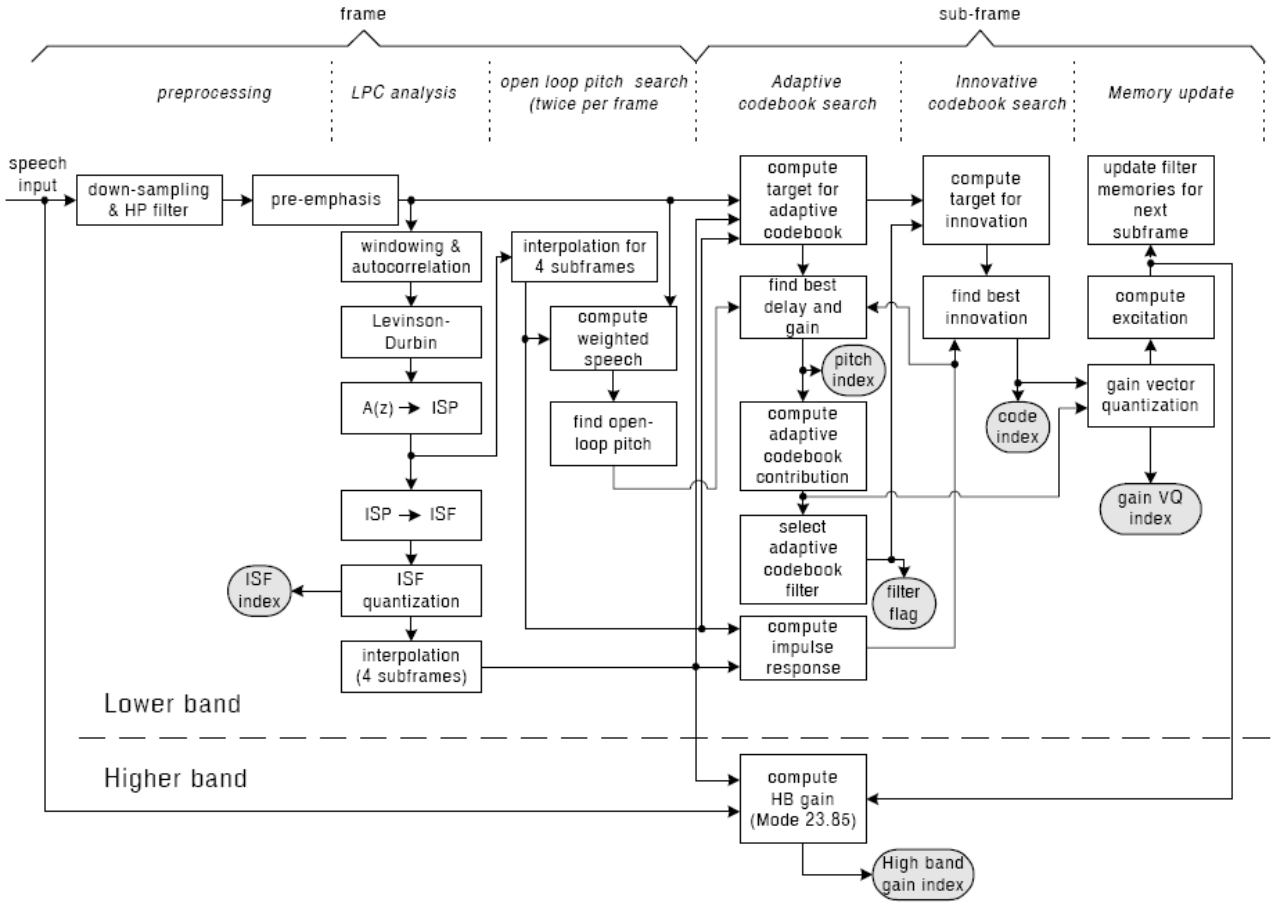
- The fundamental entity is Actor.
- Typical structure block of an Actor.



AMR-WB

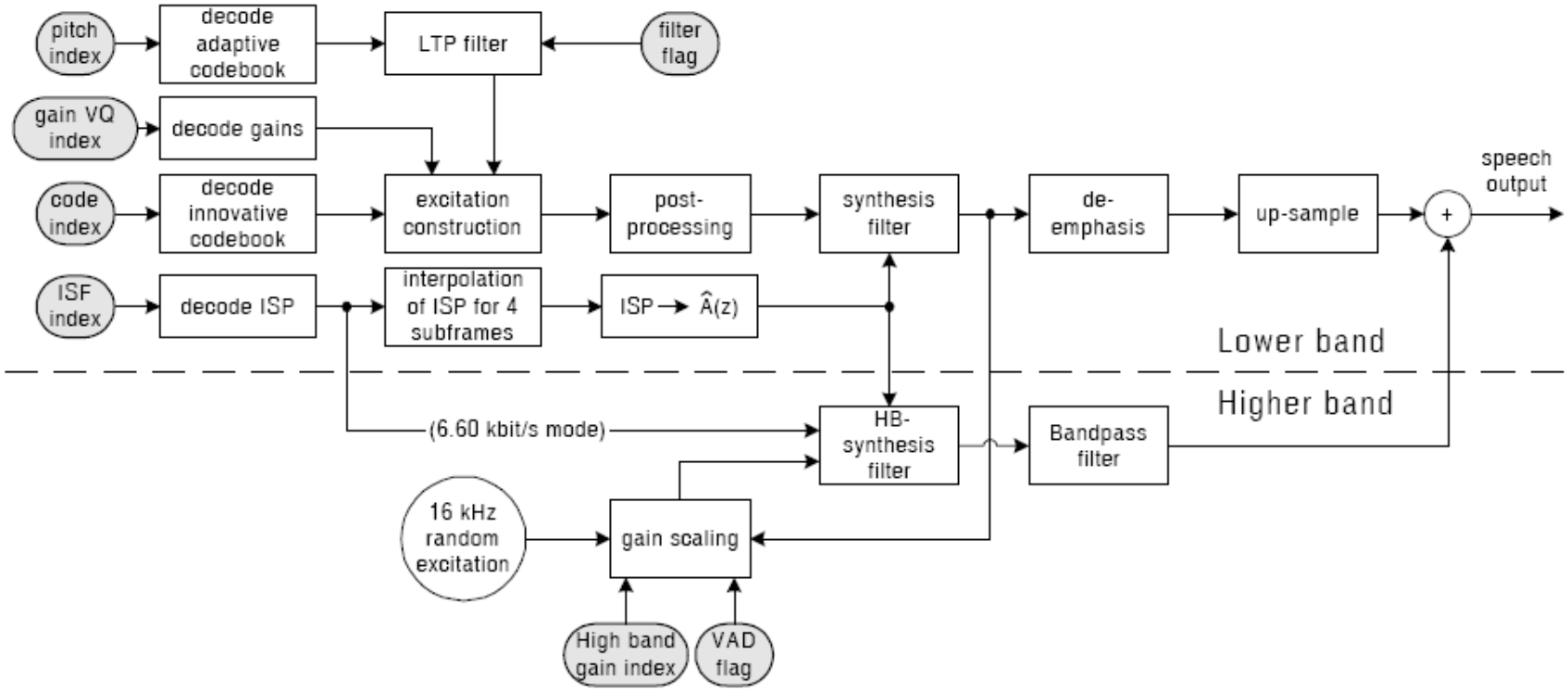
- **Adaptive Multi Rate – WideBand** (**AMR-WB**) is a speech coding standard developed using ACELP (**Algebraic code excited linear prediction**). The codec provides excellent speech quality due to wider speech bandwidth of 50–7000 Hz.
- AMR-WB operates with 9 different bit rates
- All modes are sampled at 16 kHz (using 14 bit resolution) and processed at 12.8 kHz.
- The bit rates are the following:
  - 6.60 kbit/s
  - 8.85 kbit/s
  - 12.65 kbit/s
  - 14.25 kbit/s
  - 15.85 kbit/s
  - 18.25 kbit/s
  - 19.85 kbit/s
  - 23.05 kbit/s
  - 23.85 kbit/s
  - 1.75 kbit/s

Detailed Block diagram of the AMR-WB ACELP encoder



B. Bessette et al, R. Salami et al, J. Rotola-Pukkila et al,* "The Adaptive Multi-Rate Wideband Speech Codec (AMR-WB)" , *IEEE Trans. On Speech and Audio Processing, 2002

Detailed block diagram of the AMR-WB ACELP decoder



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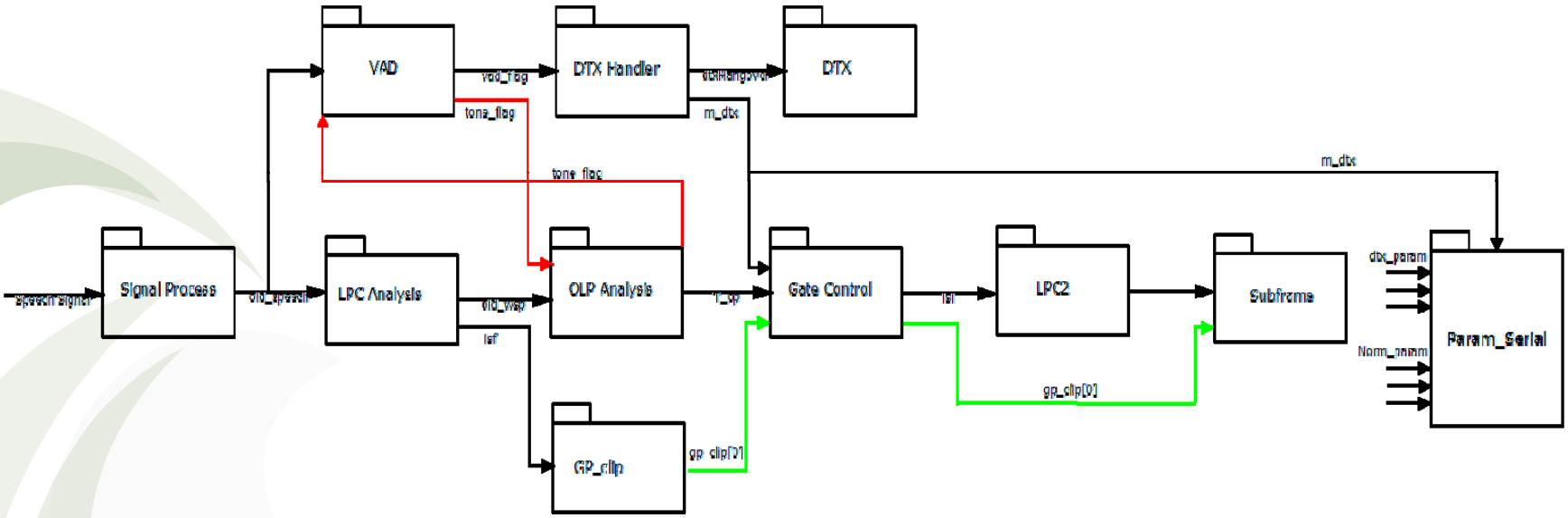
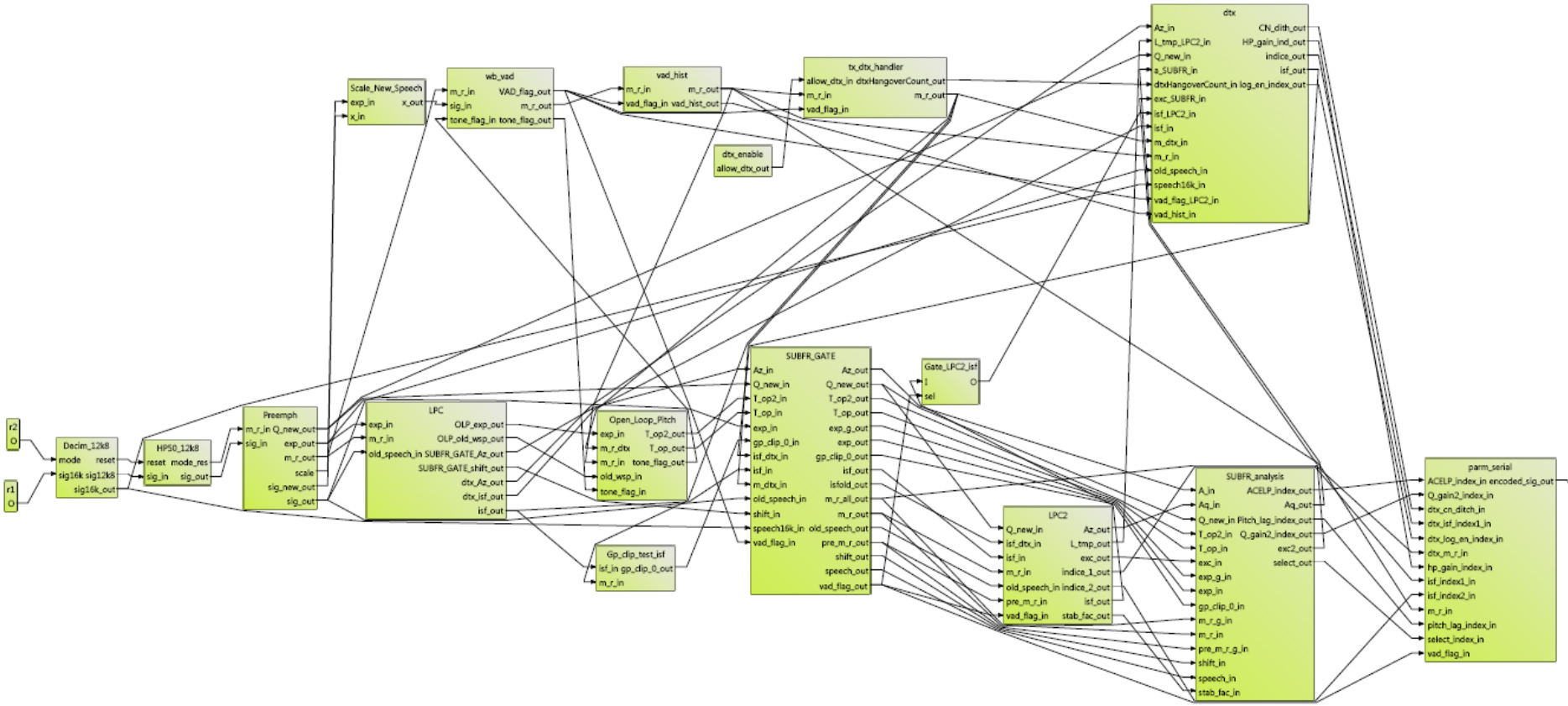
Dataflow model for the encoder




r2
O
r1
O
Decim_12k8
mode reset
sig16k sig12k8
sig16k_out
HP50_12k8
reset mode_res
sig_in sig_out
Preemph
m_r_in Q_new_out
sig_in exp_out
m_r_out
scale
sig_new_out
sig_out
LPC
exp_in OLP_exp_out
m_r_in OLP_old_wsp_out
old_speech_in SUBFR_GATE_Az_out
SUBFR_GATE_shift_out
dtx_Az_out
dtx_lsf_out
lsf_out
Open_Loop_Pitch
exp_in T_op2_out
m_r_dtx T_op_out
m_r_in tone_flag_out
old_wsp_in
tone_flag_in
Gp_clip_test_isf
lsf_in gp_clip_0_out
m_r_in
Scale_New_Speech
exp_in x_out
x_in
wb_vad
m_r_in VAD_flag_out
sig_in m_r_out
tone_flag_in tone_flag_out
vad_hist
m_r_in m_r_out
vad_flag_in vad_hist_out
dtx_enable
allow_dtx_out
tx_dtx_handler
allow_dtx_in dtxHangoverCount_out
m_r_in m_r_out
vad_flag_in
SUBFR_GATE
Az_in Az_out
Q_new_in Q_new_out
T_op2_in T_op2_out
T_op_in T_op_out
exp_in exp_g_out
gp_clip_0_in exp_out
isf_dtx_in gp_clip_0_out
isf_in isf_out
m_dtx_in isfold_out
old_speech_in m_r_all_out
shift_in m_r_out
speech16k_in old_speech_out
vad_flag_in pre_m_r_out
shift_out
speech_out
vad_flag_out
Gate_LPC2_lsf
I O
sel
dtx
Az_in CN_dith_out
L_tmp_LPC2_in HP_gain_ind_out
Q_new_in indice_out
a_SUBFR_in isf_out
dtxHangoverCount_in log_en_index_out
exc_SUBFR_in
isf_LPC2_in
isf_in
m_dtx_in
m_r_in
old_speech_in
speech16k_in
vad_flag_LPC2_in
vad_hist_in
LPC2
Q_new_in Az_out
isf_dtx_in L_tmp_out
isf_in exc_out
m_r_in indice_1_out
old_speech_in indice_2_out
pre_m_r_in isf_out
vad_flag_in stab_fac_out
SUBFR_analysis
A_in ACELP_index_out
Aq_in Aq_out
Q_new_in Pitch_lag_index_out
T_op2_in Q_gain2_index_out
T_op_in exc2_out
exc_in select_out
exp_g_in
exp_in
gp_clip_0_in
m_r_g_in
m_r_in
pre_m_r_g_in
shift_in
speech_in
stab_fac_in
parm_serial
ACELP_index_in encoded_sig_out
Q_gain2_index_in
dtx_cn_ditch_in
dtx_isf_index1_in
dtx_log_en_index_in
dtx_m_r_in
hp_gain_index_in
isf_index1_in
isf_index2_in
m_r_in
pitch_lag_index_in
select_index_in
vad_flag_in
w1
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Design Difficulties & Problems.

- Frame Dependencies affecting Pipelining.
- SubFrame Dependences affecting Parallel Computing.

Status

- Implementation work finished
- Report being written
- Presentation in January 2010