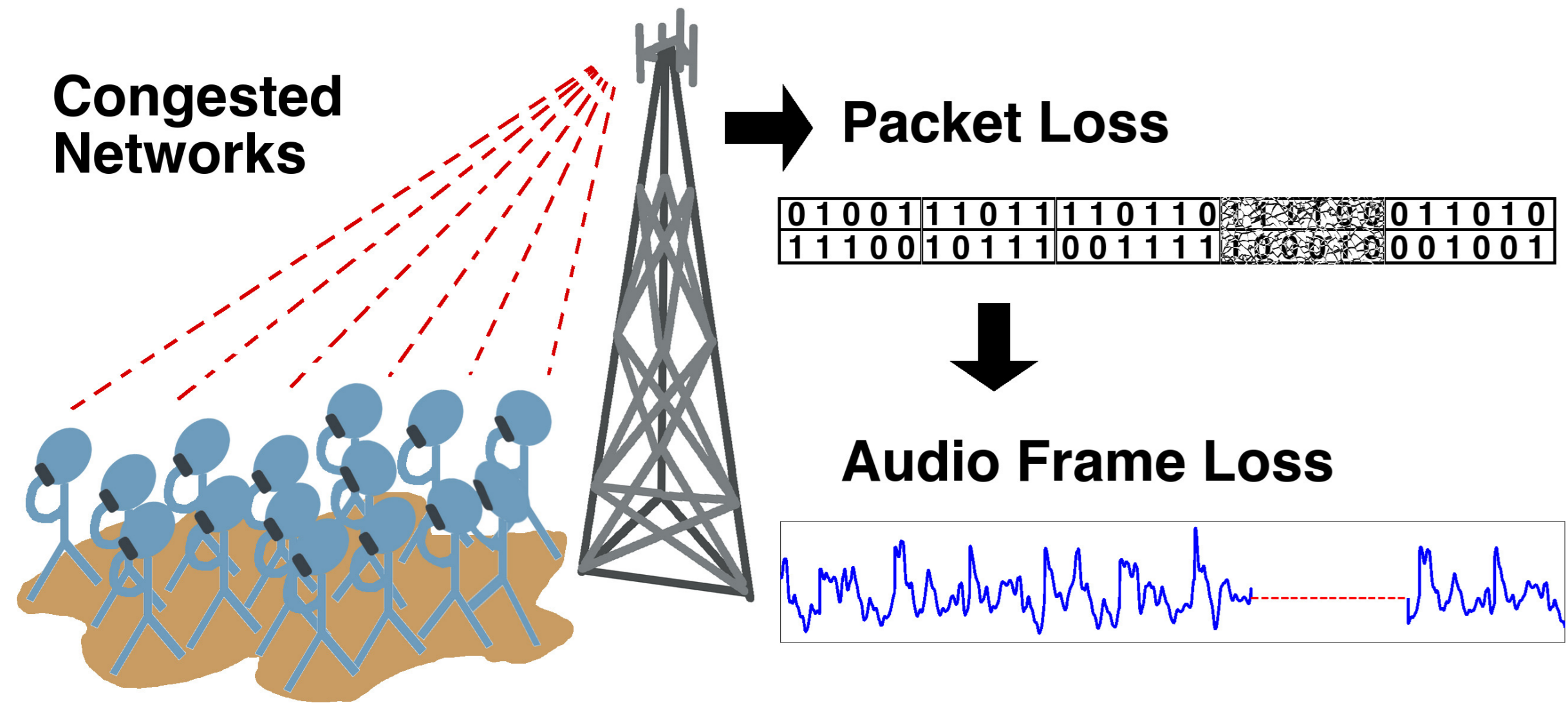
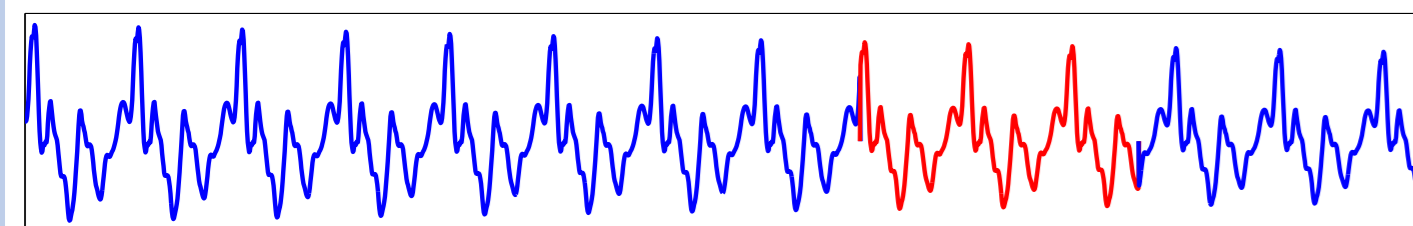


## Why Frame Loss Concealment (FLC) ?



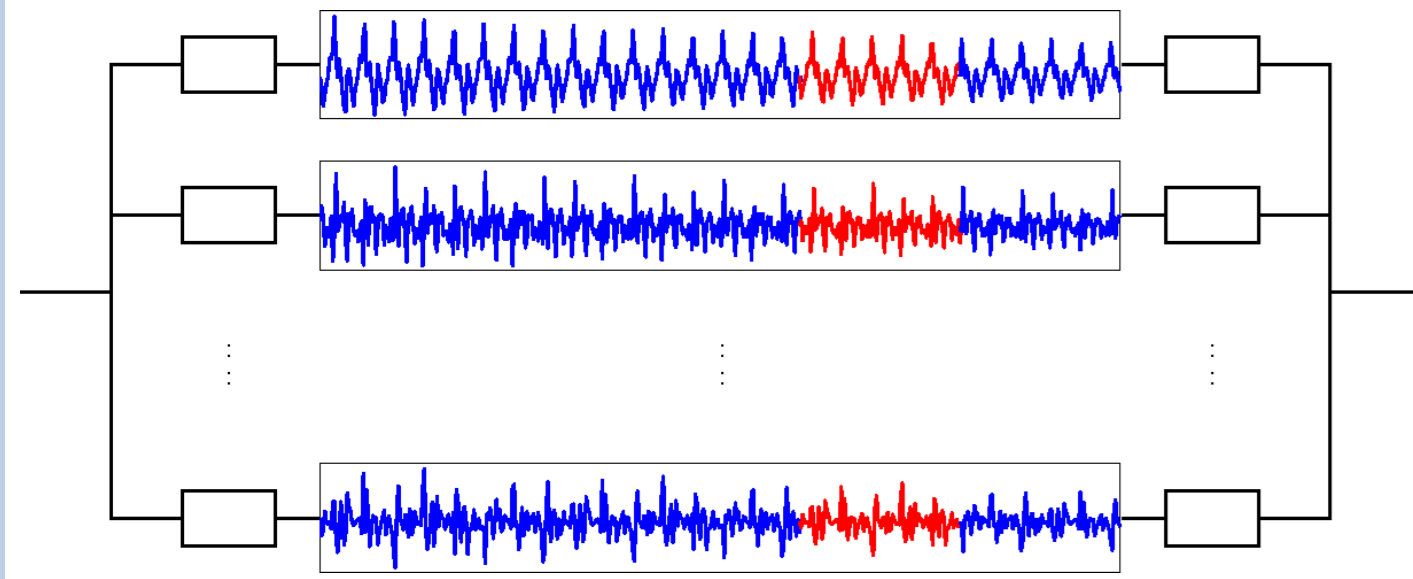
## Existing FLC techniques

### Waveform repetition



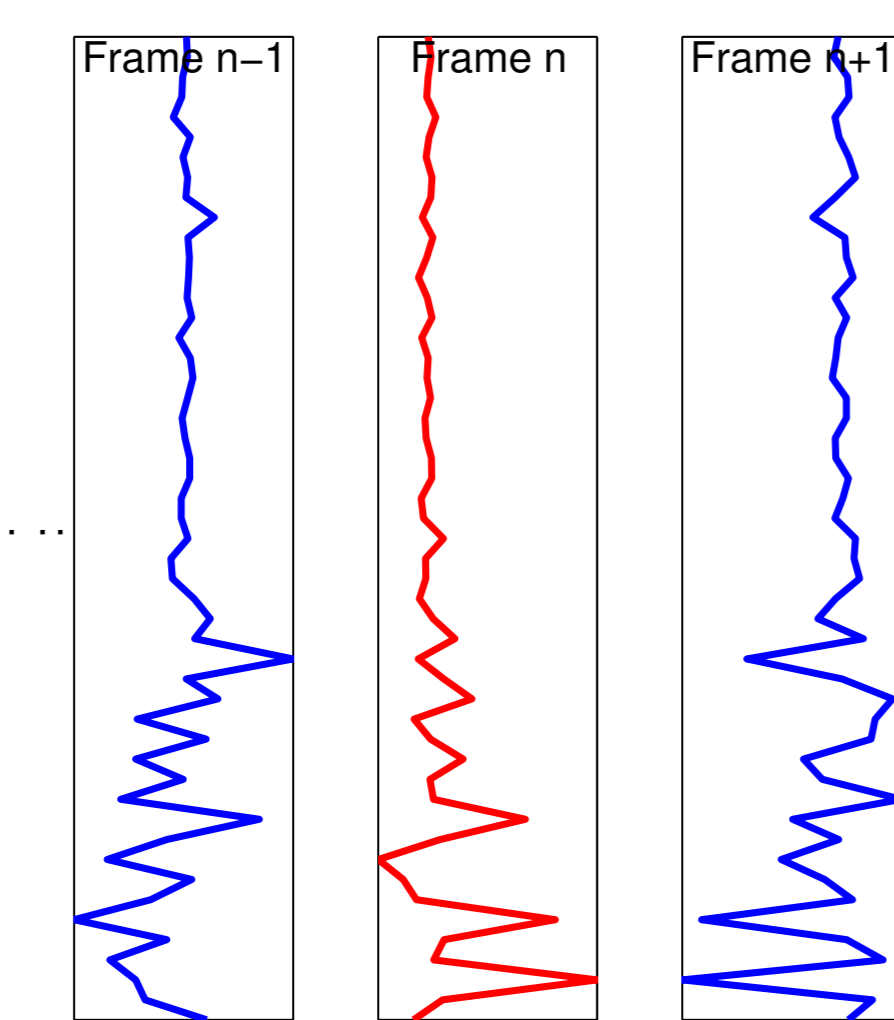
**X** Ineffective for most audio signals as they are polyphonic.

### Subband domain linear prediction



**X** Ineffective for long lost blocks.

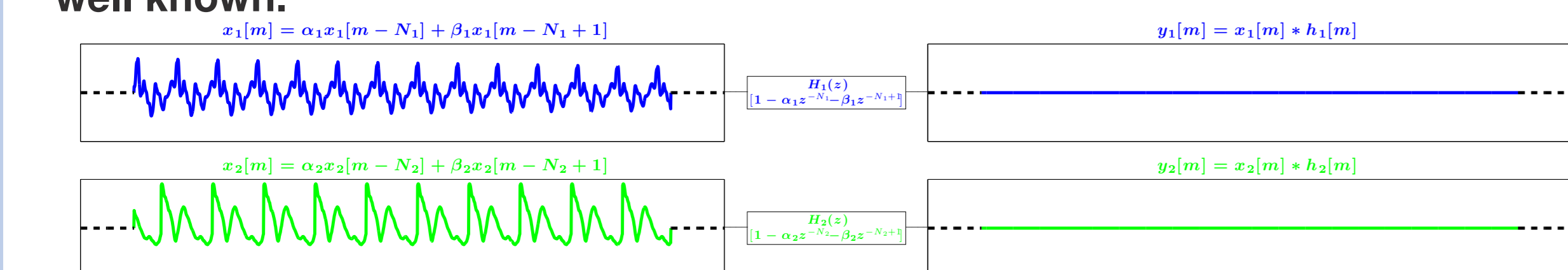
### MDCT domain tonal interpolation



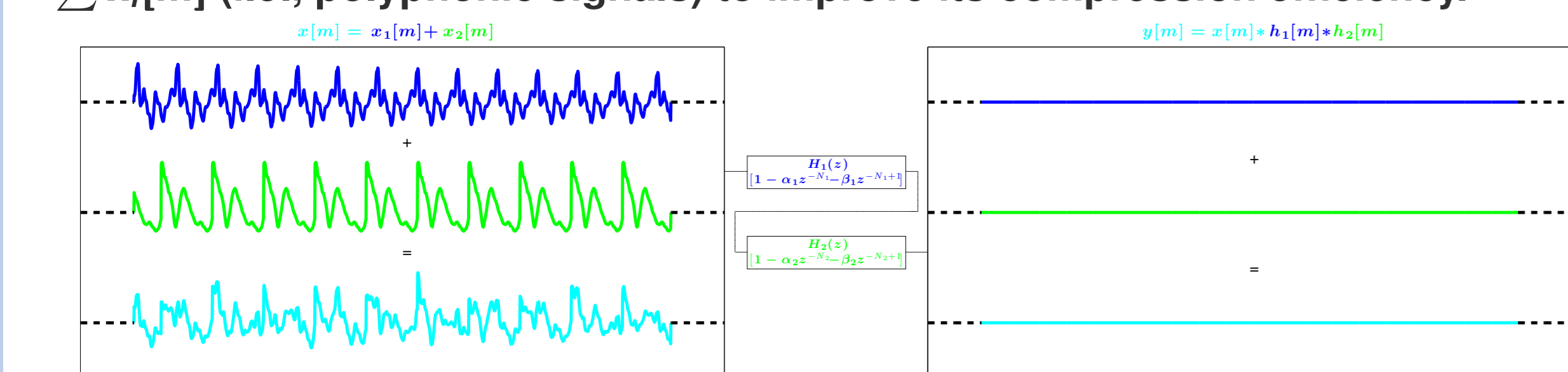
**X** Ineffective when MDCT fails to resolve individual tones.

## Cascaded Long Term Prediction (CLTP)

- Predicting periodic signals such as  $x_i[m] = \alpha_i x_i[m - N_i] + \beta_i x_i[m - N_i + 1]$  is well known.



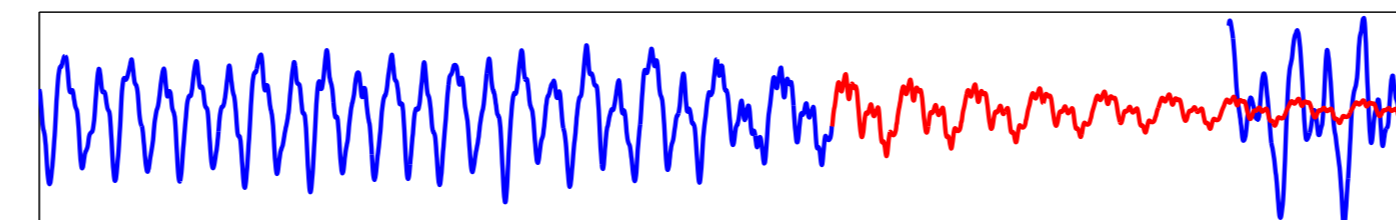
- We recently introduced CLTP for mixture of such periodic signals  $x[m] = \sum x_i[m]$  (i.e., polyphonic signals) to improve its compression efficiency.



- Here we propose employing a CLTP synthesis filter, suitably adapted for FLC by utilizing the past and even the usually available future samples effectively.

## Cascaded Long Term Prediction for Frame Loss Concealment

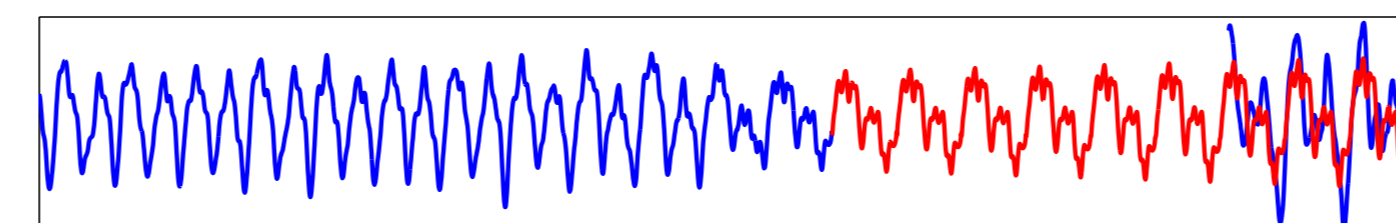
- Preliminary parameters are estimated from the past samples via a recursive technique
  - Parameters of  $j$ th filter  $(1 - \alpha_j z^{-N_j} - \beta_j z^{-N_j+1})$  are estimated in the residue of filtering with all the others  $\prod_{v, v \neq j} (1 - \alpha_v z^{-N_v} - \beta_v z^{-N_v+1})$ , via the well known technique for LTP.
  - Each filter in the cascade is estimated this way in a loop until convergence.
- Using only the past samples for the filter parameter estimate doesn't explain future samples correctly



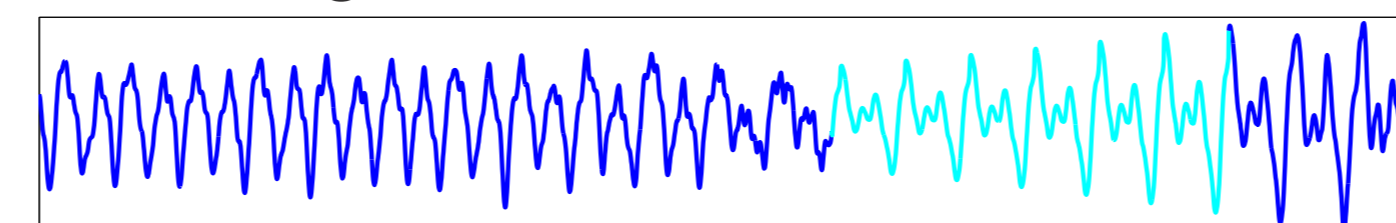
- So CLTP filter updated with multiplicative gain factors

$$H_c(z) = \prod_{i=0}^{P-1} (1 - G_i(\alpha_i z^{-N_i} + \beta_i z^{-N_i+1})).$$

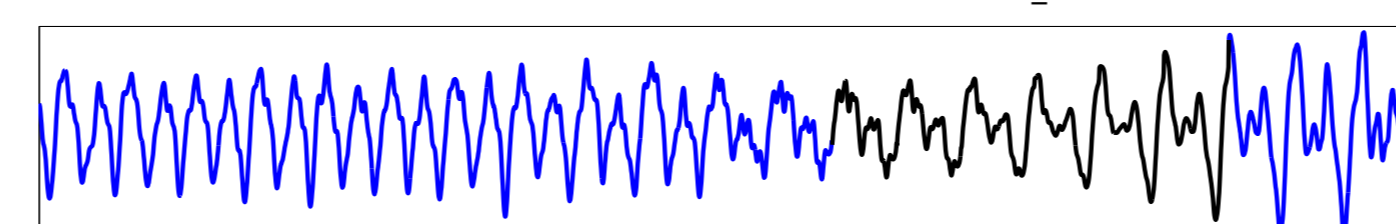
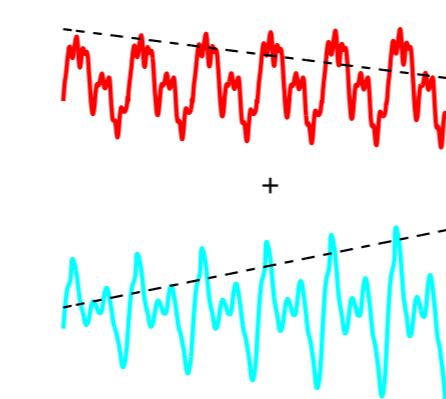
- The gain factors are adjusted to minimize squared prediction error in the future samples.
- As cost function has complex dependency on these factors, a generic quasi-Newton optimization called L-BFGS method is used along with backtracking line search for step sizes.



- Simply predicting from past samples doesn't ensure smooth transition into the available future samples.
- Thus lost frame samples are predicted in reverse direction from future samples with different set of CLTP gain factors.

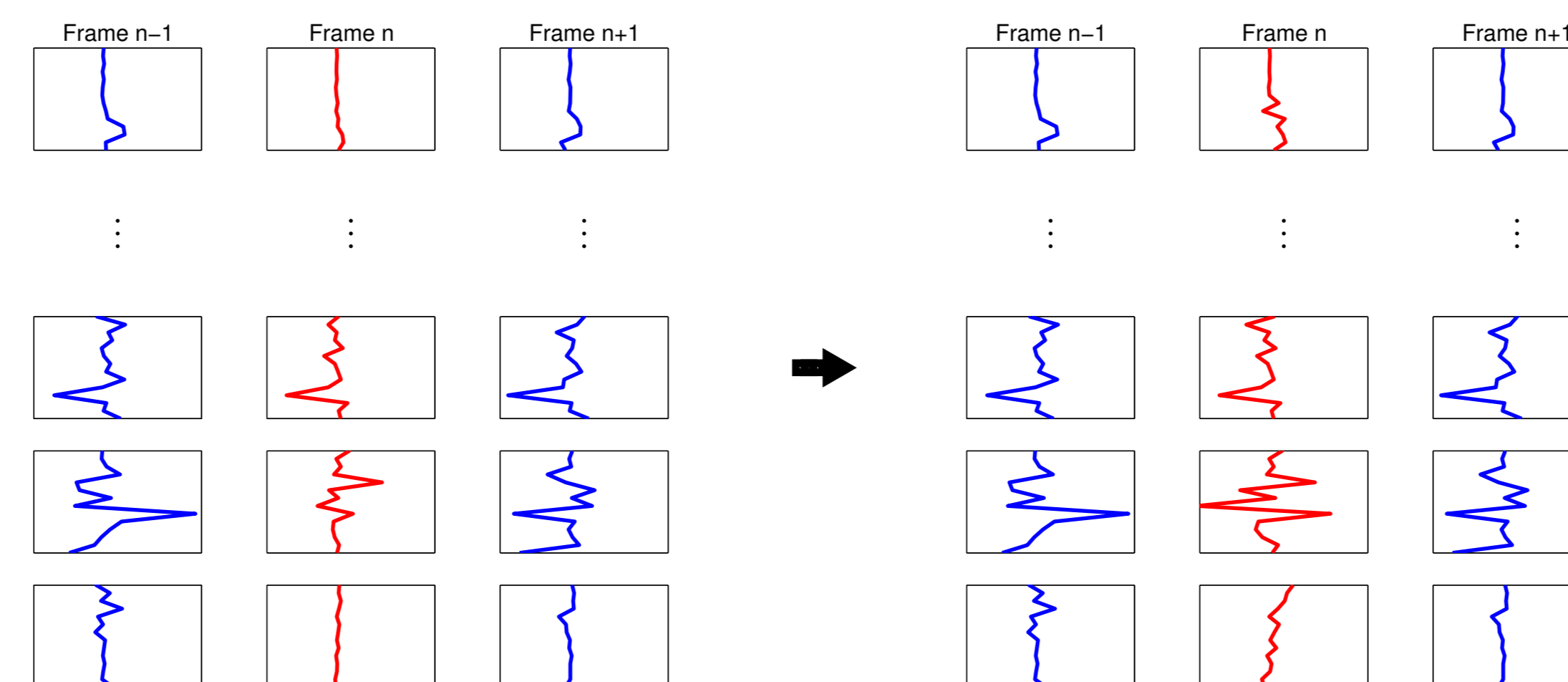


- Final reconstruction of lost frame is a weighted average of predicted samples in each direction.



- For use in MPEG AAC, the reconstructed frame is transformed to MDCT domain and energy smoothing performed in each band  $l$ , via a gain factor given as,

$$f[l] = \begin{cases} \sqrt{\frac{e_{n-1}[l]e_{n+1}[l]}{e_n[l]}}, & \text{if } \frac{e_n[l]}{\sqrt{e_{n-1}[l]e_{n+1}[l]}} > T \text{ or } \frac{e_n[l]}{\sqrt{e_{n-1}[l]e_{n+1}[l]}} < 1/T, \\ 1, & \text{otherwise.} \end{cases}$$

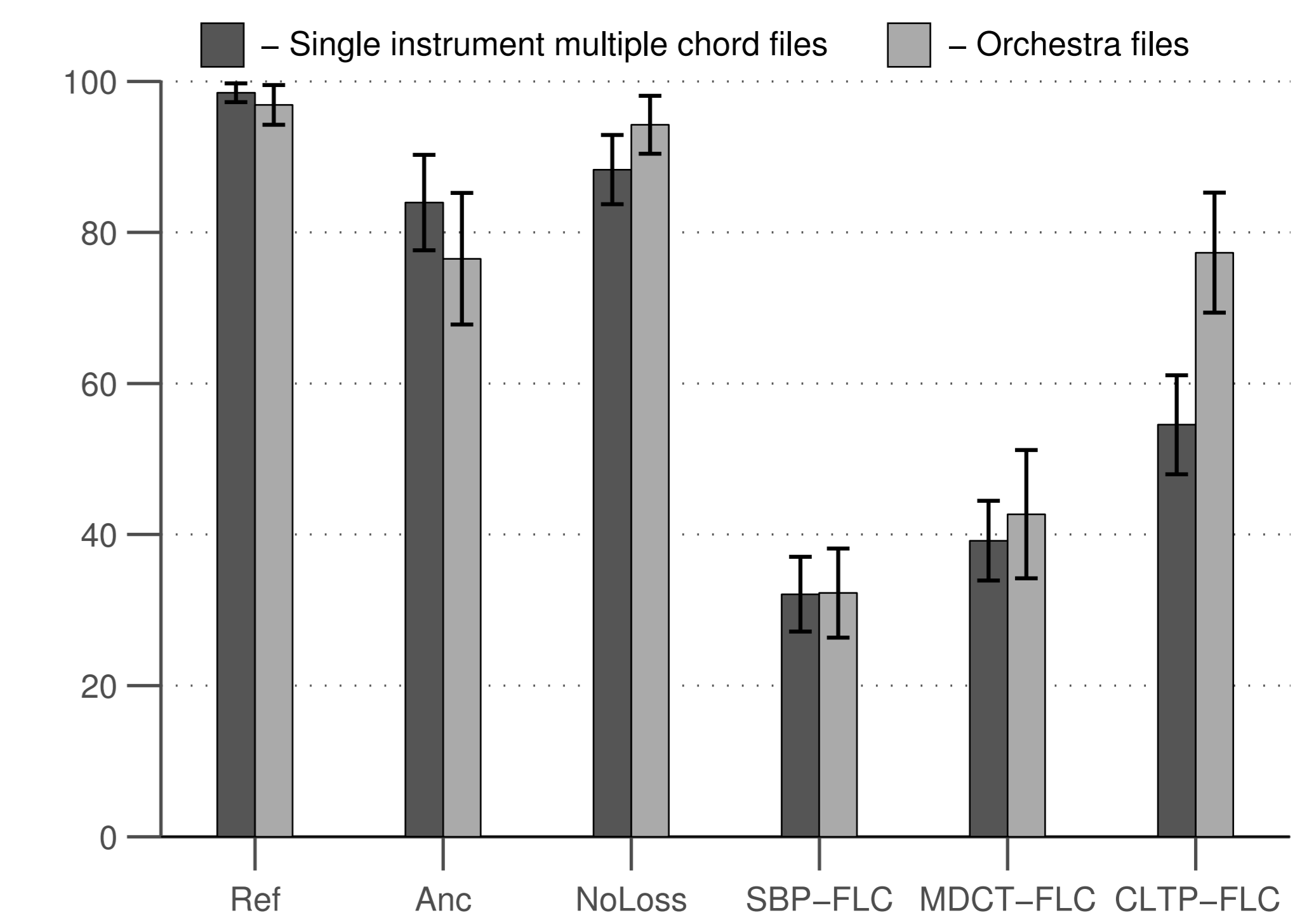


## Evaluations

- MPEG reference AAC-LD encoder used to generate 64 kbps bitstreams and the following decoders compared,
  - Reference decoder with no frame loss.
  - Reference decoder with subband domain linear prediction based FLC (SBP-FLC).
  - Reference decoder with MDCT domain tonal interpolation FLC (MDCT-FLC).
  - Reference decoder with the proposed CLTP based FLC (CLTP-FLC).
- Testing data-set: 6 audio files, 4s each, mono, 44.1/48 kHz.
- Frame loss was at the rate of 10% and random.
- Objective evaluation results of Segmental SNR in dB.

Filename	SBP-FLC	MDCT-FLC	CLTP-FLC
Piano	-3.16	-0.67	5.10
Guitar	-1.95	0.19	7.15
Harp	-3.59	-1.77	3.80
Bells	-2.08	0.06	4.26
Mfv	2.27	0.34	11.53
Mozart	-2.03	1.22	8.4
Average	-1.76	-0.11	6.71 (+6.82)

- Subjective evaluation results of MUSHRA listening tests (16 listeners, plots with average and 95% confidence interval).



## Conclusions

- Currently used FLC techniques sub-optimal for polyphonic audio signals.
- Bidirectional cascaded LTP proposed for significantly improved FLC, which takes into account all the available information.
- Subjective and objective evaluations substantiate these improvements.
- Future directions include developing low complexity variant and handling burst frame losses.