

GCT535: Sound Technology for Multimedia

# Time Scale Modification and Pitch Shifting



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Culture Technology

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# Outlines

- Understanding the algorithms to change pitch or length (or time-scale) of audio waveforms
  - Resampling
  - Time-scale modification (or time stretching)
  - Pitch shifting

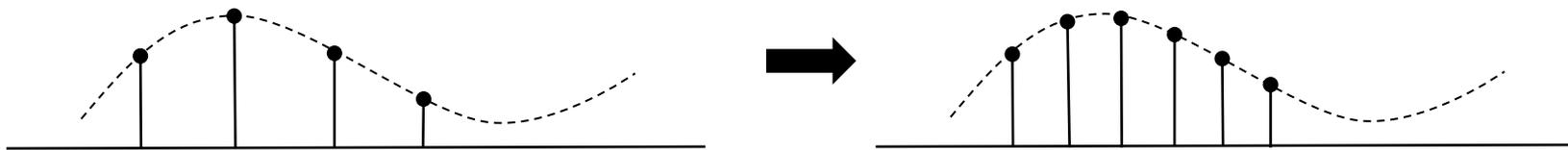
# Resampling (Playback Rate Conversion)

- “Playback rate” is not necessarily equal to the recording rate
- Adjusting the playback rate given the recorded audio change the tone
  - Sliding tapes on the magnetic header in a variable speed
  - Speeding down: “monster-like”
  - Speeding up: “chipmunk-like”
  - It can be even negative rate: reverse playback
- Demo
  - <https://musiclab.chromeexperiments.com/Voice-Spinner>



# Playback Rate Conversion (Resampling)

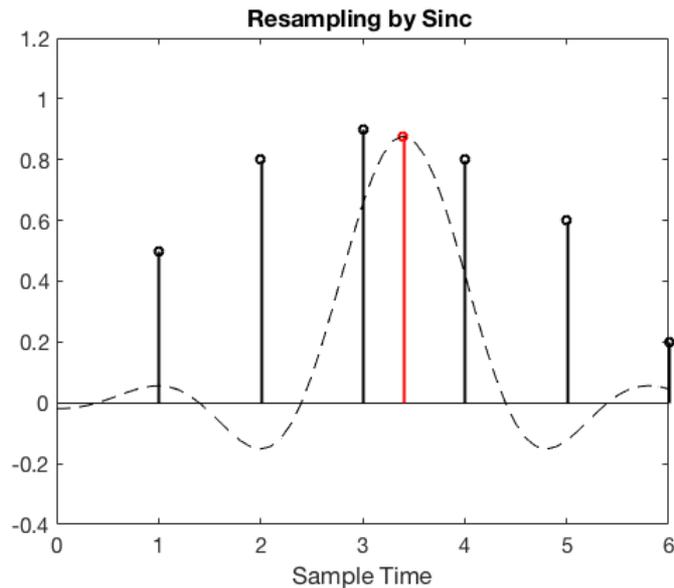
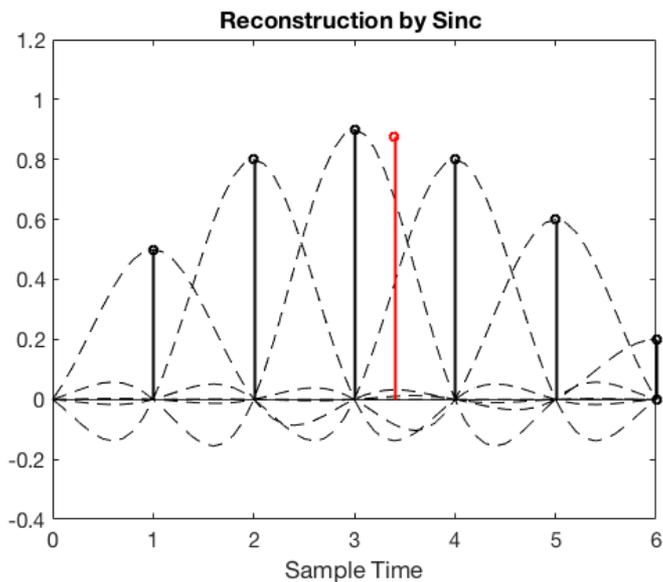
- Reconstruct the original signal and sample it with a new rate



- For a digital system with a constant playback rate
  - **Up-sampling** makes the original sound **played slower**
  - **Down-sampling** makes the original sound **played faster**

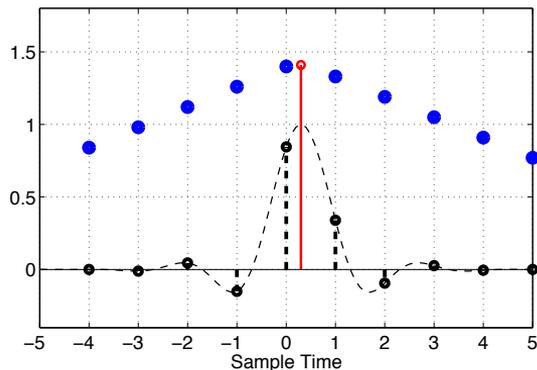
# Resampling by the Reconstruction Lowpass Filter

- As you recall from the topic of digital audio, the original signal can be reconstructed by the sinc function
  - Resampling on the reconstructed signal is equivalent to interpolation with the reconstruction filter

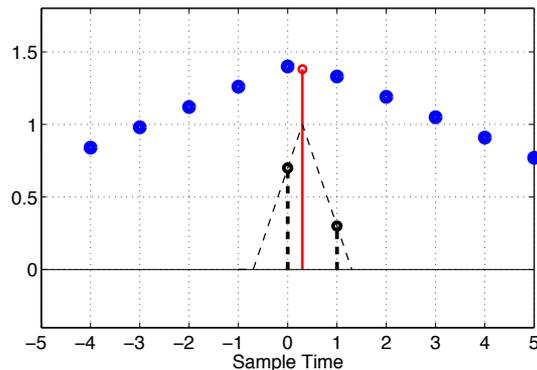


# Reconstruction Lowpass Filters (Interpolation Filters)

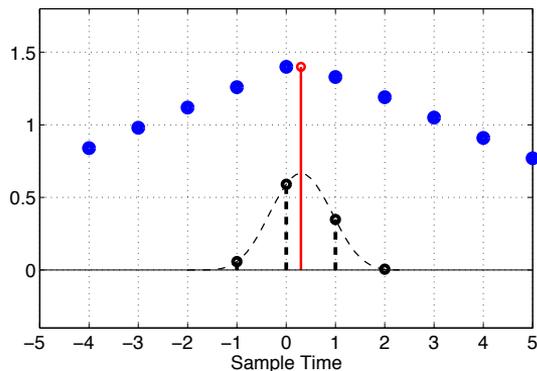
Windowed Sync



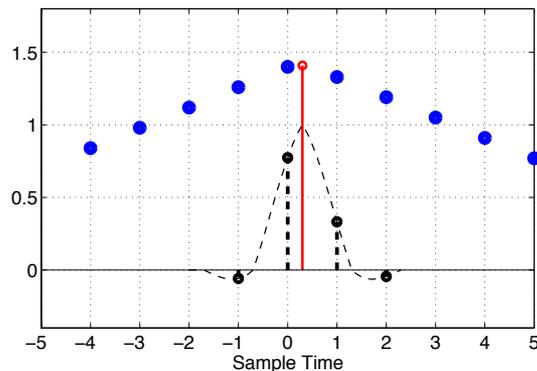
Linear Interpolation



3<sup>rd</sup> order B-spline Interpolation

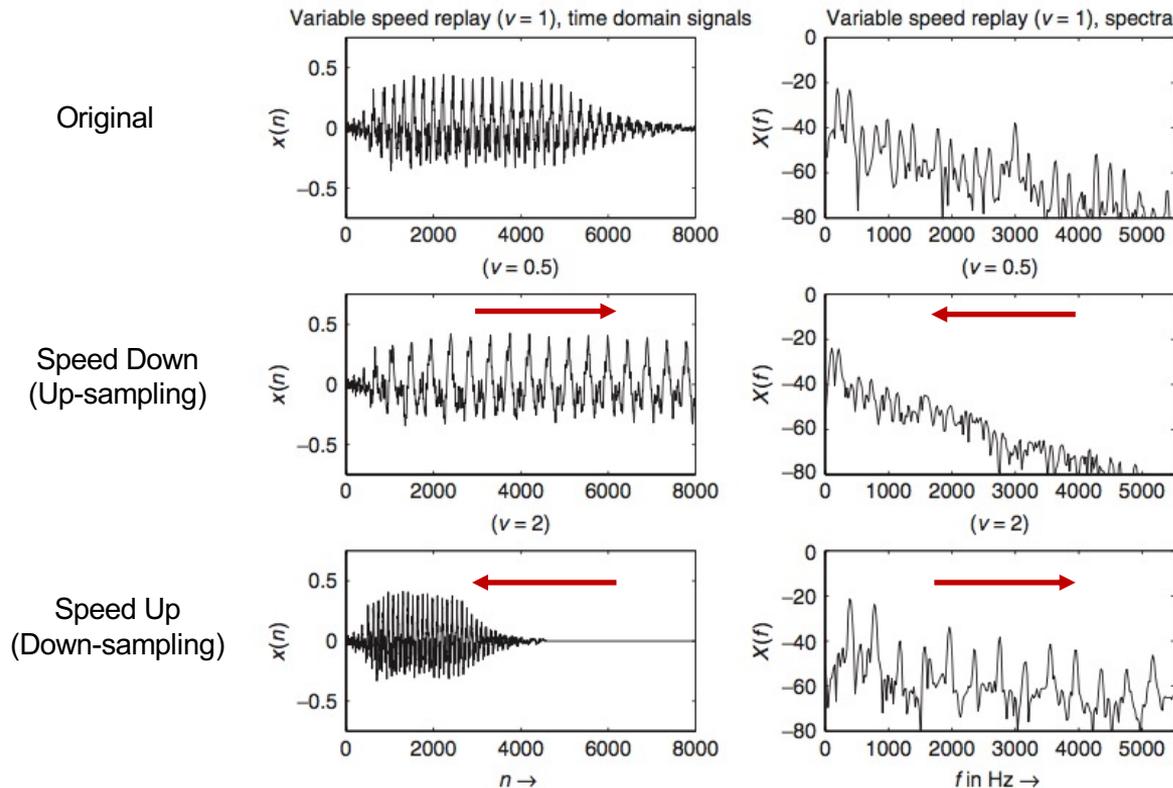


3<sup>rd</sup> order Lagrange Interpolation



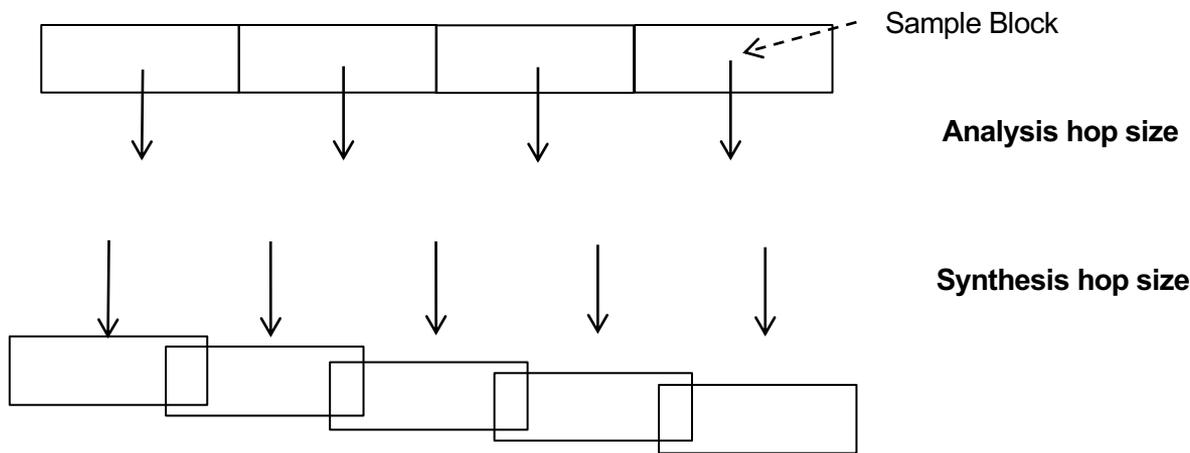
# Resampling

- Resampling changes pitch, length and timbre at the same time!



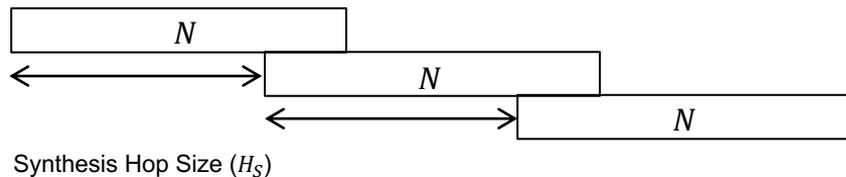
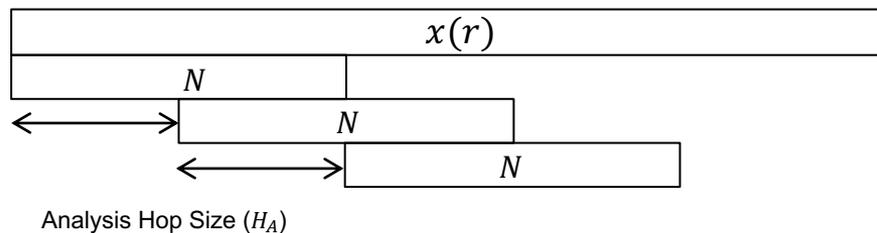
# How can we control pitch and length independently?

- The secret is processing samples in frame-level instead of sample-level
  - The waveform is locally preserved within the frame
  - Analysis hop size and synthesis hop size are distinguished



# Fundamental of Time-Scale Modification

- Analysis-Resynthesis approach in a frame-by-frame manner



$$x_m(r) = \begin{cases} x(r + mH_A), & \text{if } r \in [-\frac{N}{2} : \frac{N}{2} - 1] \\ 0, & \text{otherwise.} \end{cases}$$

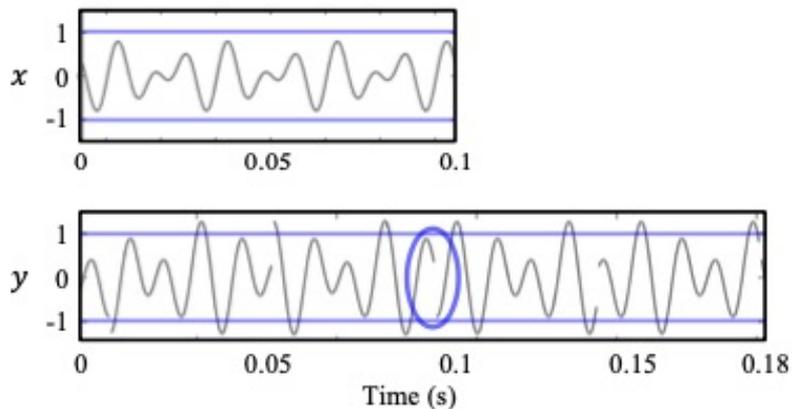
$$\mathcal{F}: x_m(r) \rightarrow y_m(r)$$

$$y(r) = \sum_{m \in \mathbb{Z}} y_m(r - mH_S)$$

- Time-stretching ratio:  $\alpha = \frac{H_S}{H_A}$  ( $H_S$ : synthesis hop size,  $H_A$ : analysis hop size)
- If  $\alpha > 1$ , increase the length, If  $\alpha < 1$ , reduce the length

# Fundamental of Time-Scale Modification (TSM)

- If the analysis frame  $x_m(r)$  is the same as the synthesis frame  $y_m(r)$ ?
  - Discontinuity at the boundary of the unmodified frames
  - Overlapping of the synthesis frame changes the amplitude



Time-scale modification with  $\alpha=1.8$  [Driedger and Müller, 2016]

# OverLap-and-Add (OLA)

- Enforce a smooth transition between frames and compensate the amplitude change

- Applying a window function  $w$  to the analysis frame: e.g. Hann window

$$w(r) = \begin{cases} 0.5(1 - \cos(\frac{2\pi(r + N/2)}{N - 1})), & \text{if } r \in [-\frac{N}{2} : \frac{N}{2} - 1] \\ 0, & \text{otherwise.} \end{cases}$$

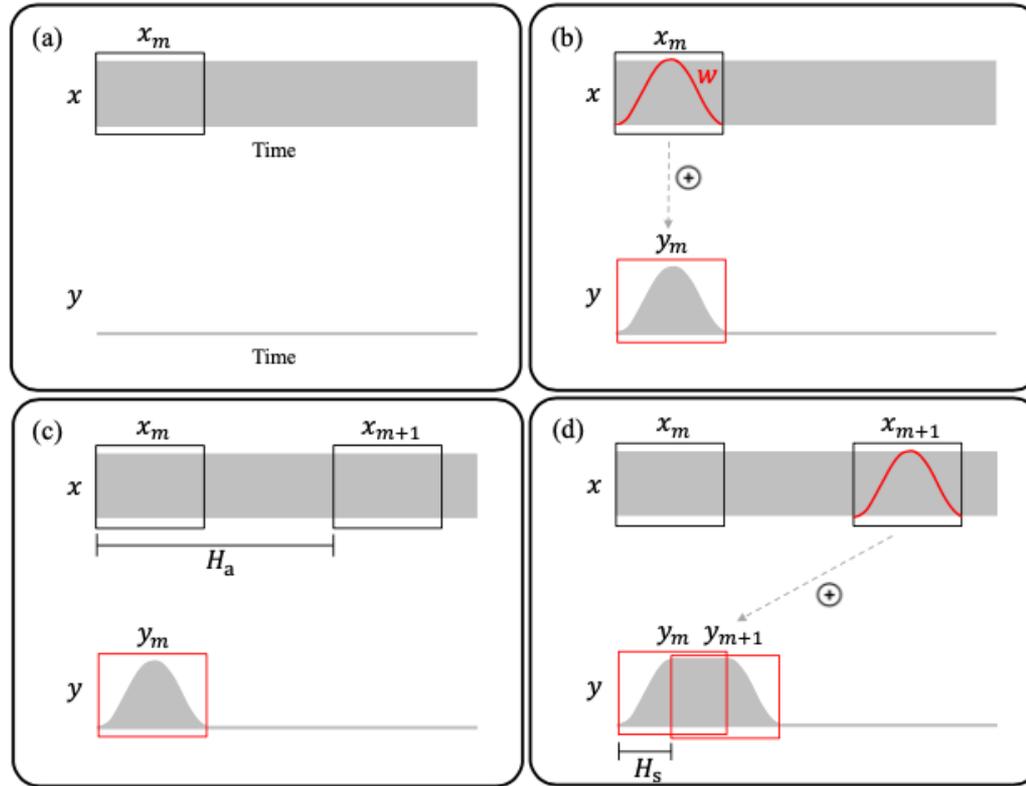
- The Hann window has the following property for all  $r \in \mathbb{Z}$

$$\sum_{n \in \mathbb{Z}} w(r - n \frac{N}{2}) = 1$$

- The synthesis frame is computed as a windowed analysis frame with the amplitude normalization

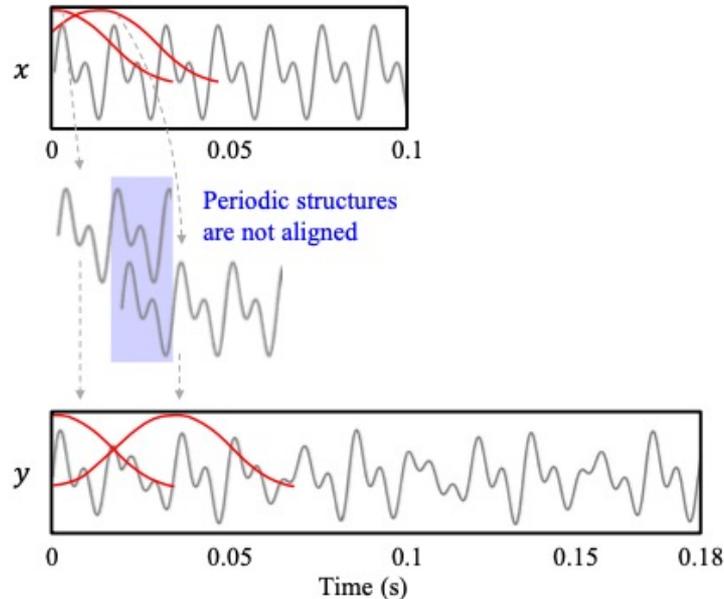
$$y_m(r) = \frac{w(r) x_m(r)}{\sum_{n \in \mathbb{Z}} w(r - nH_S)}$$

# OverLap-and-Add (OLA)



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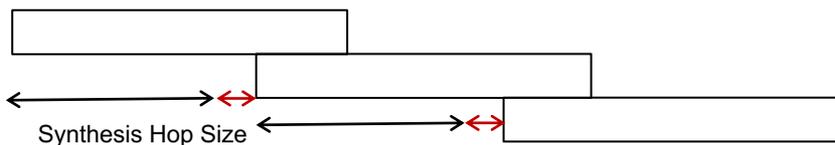
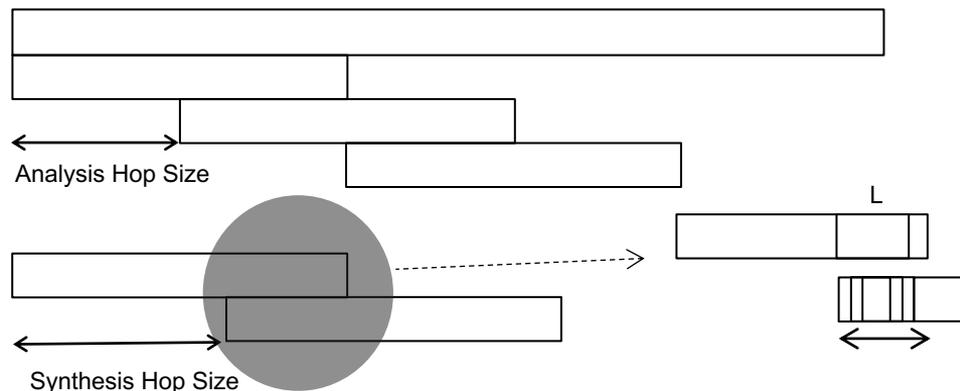
- However, OLA has a problem for periodic signals
  - They are called **phase jump artifacts**



Overlap and Add TSM [Driedger and Müller, 2016]

# Synchronized OverLap-and-Add (SOLA)

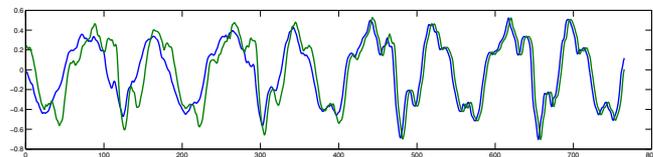
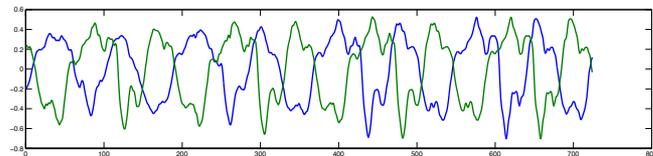
- Reduce artifacts in OLA by shifting the overlapped region such that the two adjacent frames are maximally correlated



Shift the next frame by the lag

Synchronization by cross-correlation

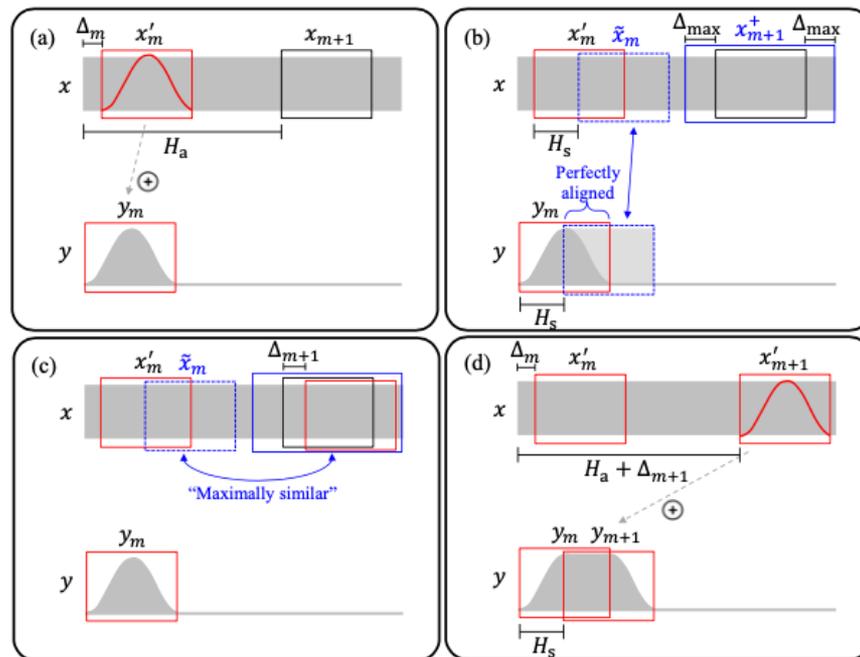
$$X_{corr}(l) = \sum_{n=0}^{n=L-1} x_1(n)x_2(n+l)$$



Find the lag ( $l$ ) where the cross correlation is maximum

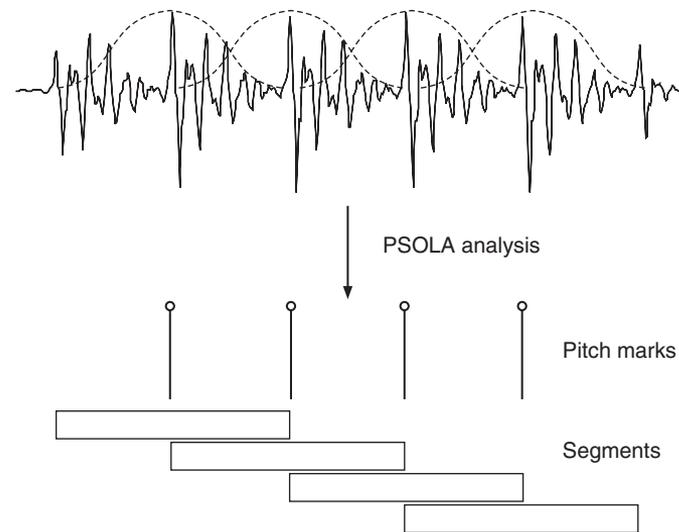
# Waveform Similarity OverLap-and-Add (WSOLA)

- A variant of SOLA that adjusts the analysis hop size



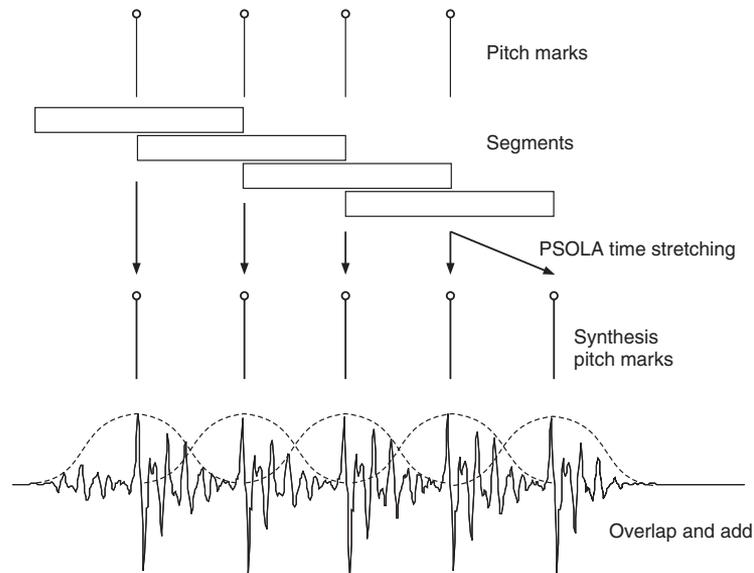
# Pitch-Synchronous OLA (PSOLA)

- The analysis and synthesis hop size is synchronized to estimated pitch
- Analysis
  - Perform block-based pitch detection and find pitch marks  $t_i$ 
    - Pitch period:  $P(t) = t_{i+1} - t_i$
  - Extract a segment centered at every pitch mark  $t_i$  using a Hanning window with length  $L_i = 2P(t_i)$  to ensure fade-in and fade-out



# Pitch-Synchronous OLA (PSOLA)

- Synthesis for time-stretching
  - For every synthesis pitch mark  $\tilde{t}_k$ , search the corresponding  $t_i$  that minimizes  $|\alpha t_i - \tilde{t}_k|$
  - Overlap and add the selected segment
    - If  $\alpha > 1$ , some segments will be repeated
    - If  $\alpha < 1$ , some segments will be discarded
  - The next synthesis pitch mark  $\tilde{t}_{k+1}$  is determined to preserve local pitch
    - $\tilde{t}_{k+1} = \tilde{t}_k + \tilde{P}(\tilde{t}_k) = \tilde{t}_k + P(t_i)$



# Pitch-Shifting

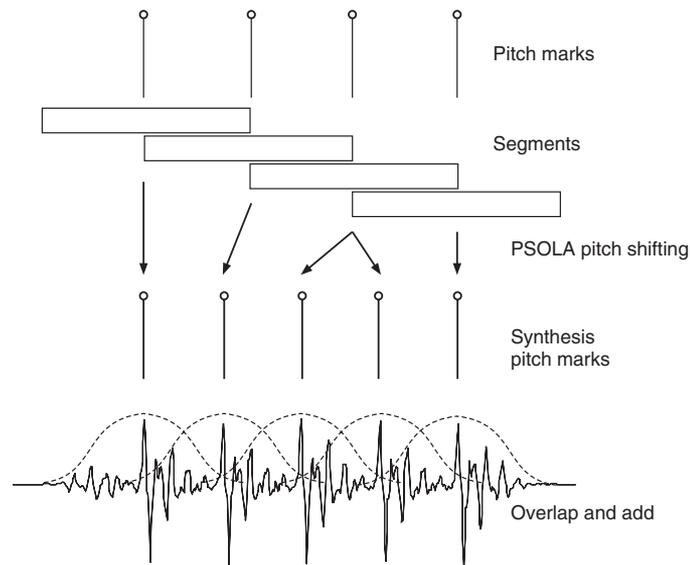
- TSM followed by Resampling
  - First, perform time-stretching with a ratio of  $\alpha$
  - Second, resampling the output with the same ratio of  $\alpha$
- Problem
  - Timbre ( i.e. formant) changes by the resampling
  - This is quite audible for human voice (e.g. speech or singing )

# Ambient Sound Design Using Paul's Extreme Sound Stretch

- Extreme time scale modification (e.g.  $\alpha = 50$ ) with spectral smoothing can transform any sound/music into a texture or ambient sound
  - <http://hypermammut.sourceforge.net/paulstretch/>
  - <https://cdm.link/2018/02/free-plug-brings-extreme-paulstretch-stretching-daw/>

# Pitch-Shifting

- PSOLA can be used for pitch-shifting
  - For every synthesis pitch mark  $\tilde{t}_k$ , search the corresponding  $t_i$  that minimizes  $|t_i - \tilde{t}_k|$
  - Overlap and add the selected segment
    - If  $\beta > 1$ , some segments will be repeated
    - If  $\beta < 1$ , some segments will be discarded
  - The next synthesis pitch mark  $\tilde{t}_k$  is determined to preserve local pitch
    - $\tilde{t}_{k+1} = \tilde{t}_k + \tilde{P}(\tilde{t}_k) = \tilde{t}_k + P(t_i)/\beta$
  - It is possible to combine the time-stretching (with the term  $|\alpha t_i - \tilde{t}_k|$ ) and pitch-shifting
  - This preserves the formant of the input sound!



# Resources

- PyTSMMod
  - Time-scaling modification code using WSOLA (waveform similarity OLSA)
  - <https://github.com/KAIST-MACLab/PyTSMMod>