

## Spectral and temporal modifications

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In this practical work, you will implement the PSOLA method for the analysis/synthesis of speech signals. This method will be tested on signals that can be downloaded on the TSIA 206 Moodle. These signals are sampled at  $F_s$ . You can load them with Matlab e.g. by typing up `load aeiou`; they will then be stocked in variable `s`. To listen to them, you can type up `soundsc(s,Fs)`. In Python, you can use the provided notebook template `template-TP-modifications.ipynb` that you can download on the website.

## 1 Extraction of the analysis marks

Firstly, you will program the following function

```
function A = AnalysisPitchMarks(s,Fs)
```

which extracts the analysis marks. The arguments `s` and `Fs` respectively are the signal to be analyzed and the sampling frequency. The returned matrix `A` will contain the times and pitches corresponding to each analysis mark. More precisely, `A` will be formed of three rows, such that  $A(1,n) = t_a(n)$  is the time corresponding to the  $n^{\text{th}}$  analysis mark ( $t_a(n) \in \mathbb{N}$  is expressed in number of samples),  $A(2,n) = \text{voiced}(n)$  is a Boolean which indicates whether the signal is voiced or unvoiced in the neighborhood of this mark, and  $A(3,n) = P_a(n) \in \mathbb{N}$  describes the pitch corresponding to the same mark (i.e. the period expressed in number of samples) in the voiced case, or equals  $10\text{ms} \times F_s$  in the unvoiced case.

To do so, you will need a pitch estimator. In order to spare time, you can use function `period.m`, whose Matlab code is provided with the example signals, and whose Python code is included in the notebook template `template-TP-HR.ipynb`. This function requires two arguments: a short term signal `x` extracted from `s`, and the sampling frequency `Fs` (the other arguments are optional), and returns a couple `[P, voiced]` where `voiced` is a Boolean which indicates whether `x` is voiced or non, and  $P \in \mathbb{N}$  is the period expressed in number of samples in the voiced case, or equals  $10\text{ms} \times F_s$  in the unvoiced case.

Let us now detail how to determine the analysis marks. For the sake of simplicity, we will not try to align the mark  $t_a(n)$  on the beginning of a glottal pulse. To compute  $P_a(n)$  and  $t_a(n)$ , we proceed by recursion on  $n \geq 1$ :

- extraction of a sequence `x` that starts at time  $t_a(n-1)$ , and whose duration is equal to  $2.5 P_a(n-1)$ ;
- computation of  $P_a(n)$  and  $\text{voiced}(n)$  by means of function `period`;
- computation of  $t_a(n) = t_a(n-1) + P_a(n)$ .

The algorithm will be initialized by setting  $t_a(0) = 1$  (in Matlab) or  $t_a(0) = 0$  (in Python) and  $P_a(0) = 10\text{ms} \times F_s$ .

## 2 Synthesis and modification of the temporal and spectral scales

To perform the synthesis of the signal, we must start by defining the synthesis marks. They will be stocked in a matrix `B` formed of two rows, such that  $B(1,k) = t_s(k)$  is the time corresponding to the  $k^{\text{th}}$  synthesis mark, and  $B(2,k) = n(k)$  is the index of the analysis mark corresponding to this same synthesis mark. To start, you can perform a synthesis without modification, by setting:

- $B(1,:) = A(1,:)$ ;
- $B(2,:) = [1, 2, 3, \dots]$ .



## 2.1 Signal synthesis

You will now program the following function

```
function y = Synthesis(s,Fs,A,B)
```

which computes the synthesis signal  $y$  from the original signal  $s$ , the sampling frequency  $F_s$ , the analysis marks stocked in matrix  $A$  and the synthesis marks stocked in matrix  $B$ . The synthesis is very simply performed by recursion on  $k \geq 1$  (vector  $y$  being initialized to the zero vector of dimension  $t_s(k_{\text{end}}) + P_a(n(k_{\text{end}}))$ ):

- extraction of a sequence  $x$  centered at  $t_a(n(k))$  and of length  $2P_a(n(k)) + 1$ ;
- windowing of  $x$  by a Hann window (Matlab function `hann` or Python function `scipy.signal.hanning`);
- overlap-add of the sequence  $x$  windowed on  $y(t_s(k) - P_a(n(k)) : t_s(k) + P_a(n(k)))$ .

## 2.2 Modification of the temporal scale

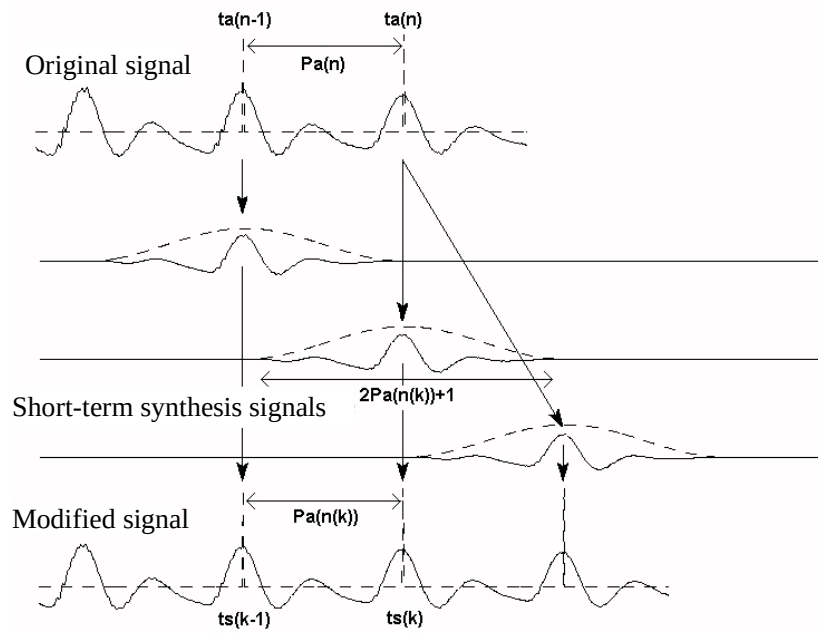


Figure 1: Modification of the temporal scale

We now want to determine the synthesis marks that will modify the temporal scale by a factor  $\alpha$ , i.e. to determine a matrix  $B$  such that the duration of the signal synthesized by function `Synthesis` is equal to that of the original signal  $s$  multiplied by  $\alpha$ . This operation will be performed by function

```
function B = ChangeTimeScale(alpha,A,Fs)
```

which computes matrix  $B$  from the factor  $\alpha$ , the analysis marks stocked in  $A$ , and the sampling frequency  $F_s$ . You can proceed by recursion on  $k \geq 1$ , by using a non-integer index  $n(k)$ :

- $t_s(k) = t_s(k-1) + P_a(\lfloor n(k) \rfloor)$ ;
- $n(k+1) = n(k) + \frac{1}{\alpha}$ .

The algorithm will be initialized by setting  $t_s(0) = 1$  and  $n(1) = 1$ . You will take care of only stocking integer values in matrix B.

### 2.3 Modification of the spectral scale

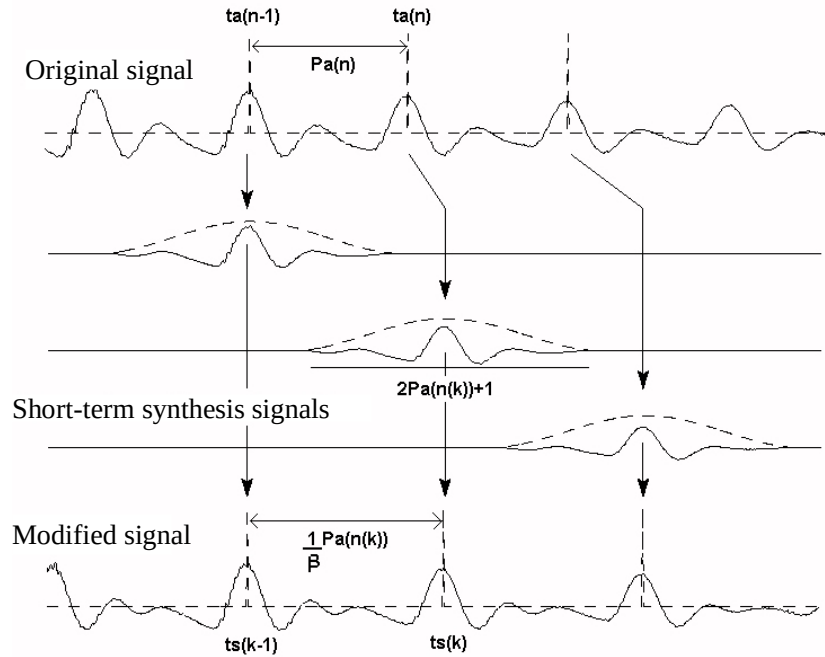


Figure 2: Modification of the spectral scale

You will now perform the dual operation of the previous one: determine the synthesis marks that will modify the spectral scale by a factor  $\beta$ , i.e. determine a matrix B such that the fundamental frequency of the signal synthesized by function `Synthesis` is equal to that of the original signal  $s$  multiplied by  $\beta$ . This operation will be performed by function

`function B = ChangePitchScale(beta,A,Fs)`

which computes matrix B from the factor  $\beta$ , the analysis marks stocked in A, and the sampling frequency Fs. As in the previous case, you can proceed by recursion on  $k \geq 1$ , by using a non-integer index  $n(k)$  and non-integer synthesis times  $t_s(k)$ , and by making the difference between the voiced and unvoiced cases:

- if the analysis mark of index  $\lfloor n(k) \rfloor$  is voiced,  $\text{scale}(k) = \frac{1}{\beta}$ , otherwise  $\text{scale}(k) = 1$ ;
- $t_s(k) = t_s(k-1) + \text{scale}(k) \times P_a(\lfloor n(k) \rfloor)$ ;
- $n(k+1) = n(k) + \text{scale}(k)$ .

Again, you will take care of only stocking integer values in matrix B.

## 2.4 Joint modification of the temporal and spectral scales

To finish, you will program a function that jointly modifies the two scales:

```
function B = ChangeBothScales(alpha,beta,A,Fs)
```

where the arguments are defined as previously. The content of this function will be almost identical to that of `ChangePitchScale`; you will just need to modify it properly.



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