Speech Analysis

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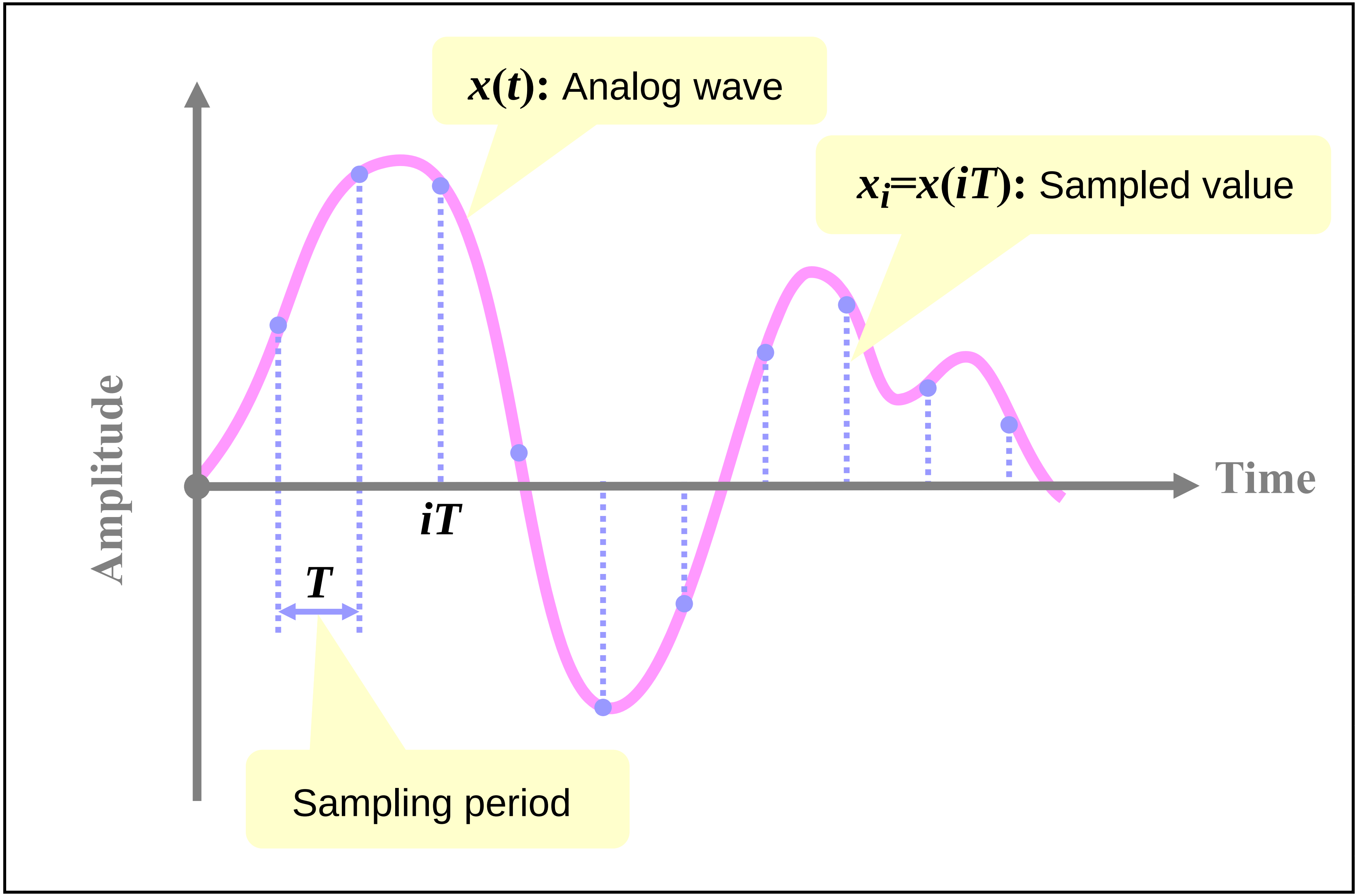
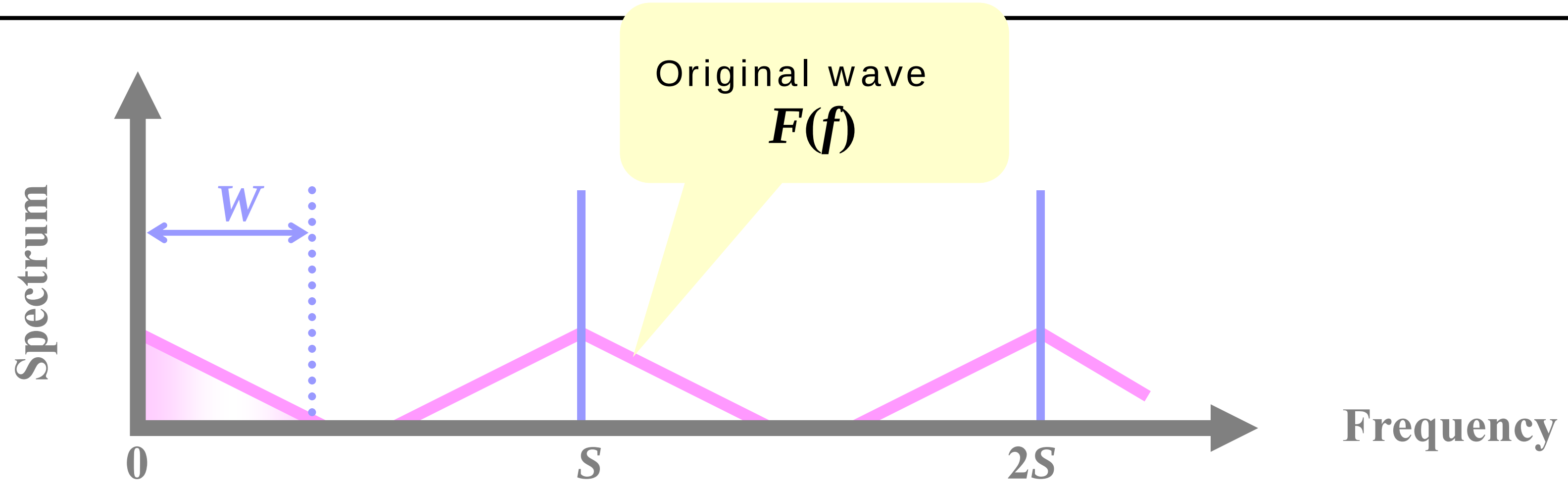
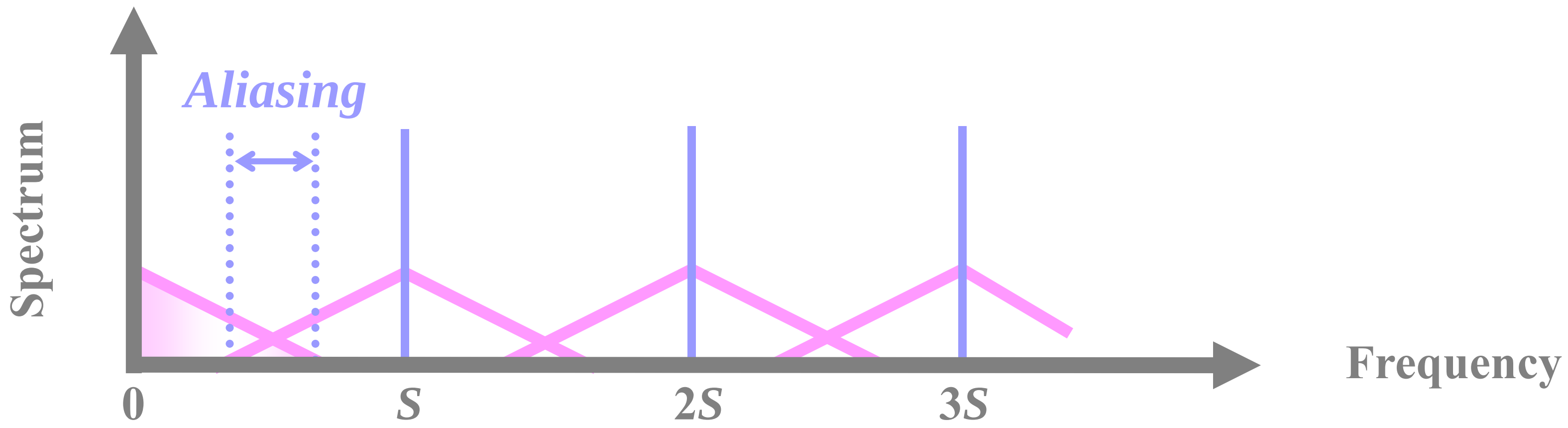


Sampling in the time domain

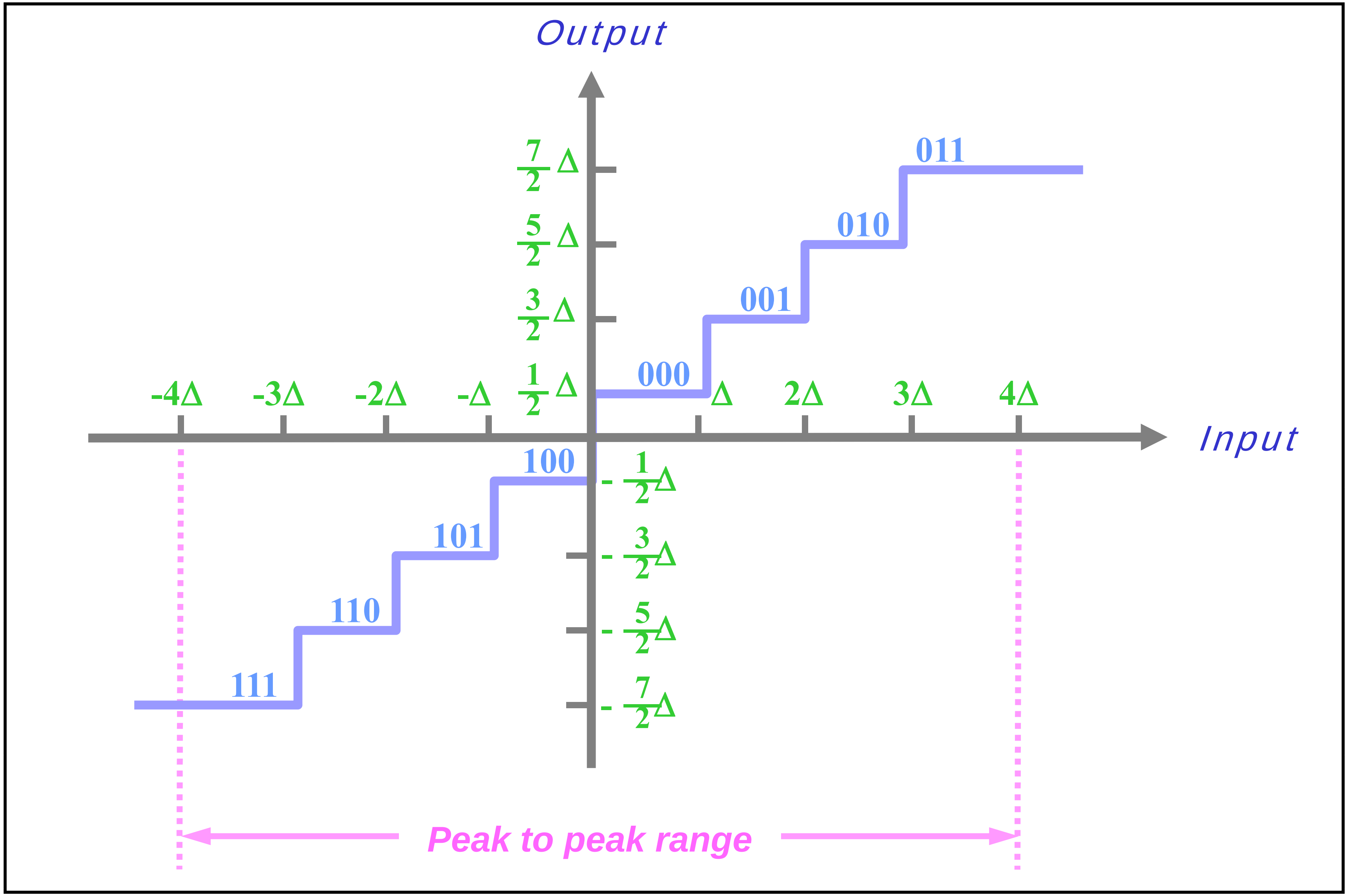


(a) Correct sampling ($S \geq 2W$)

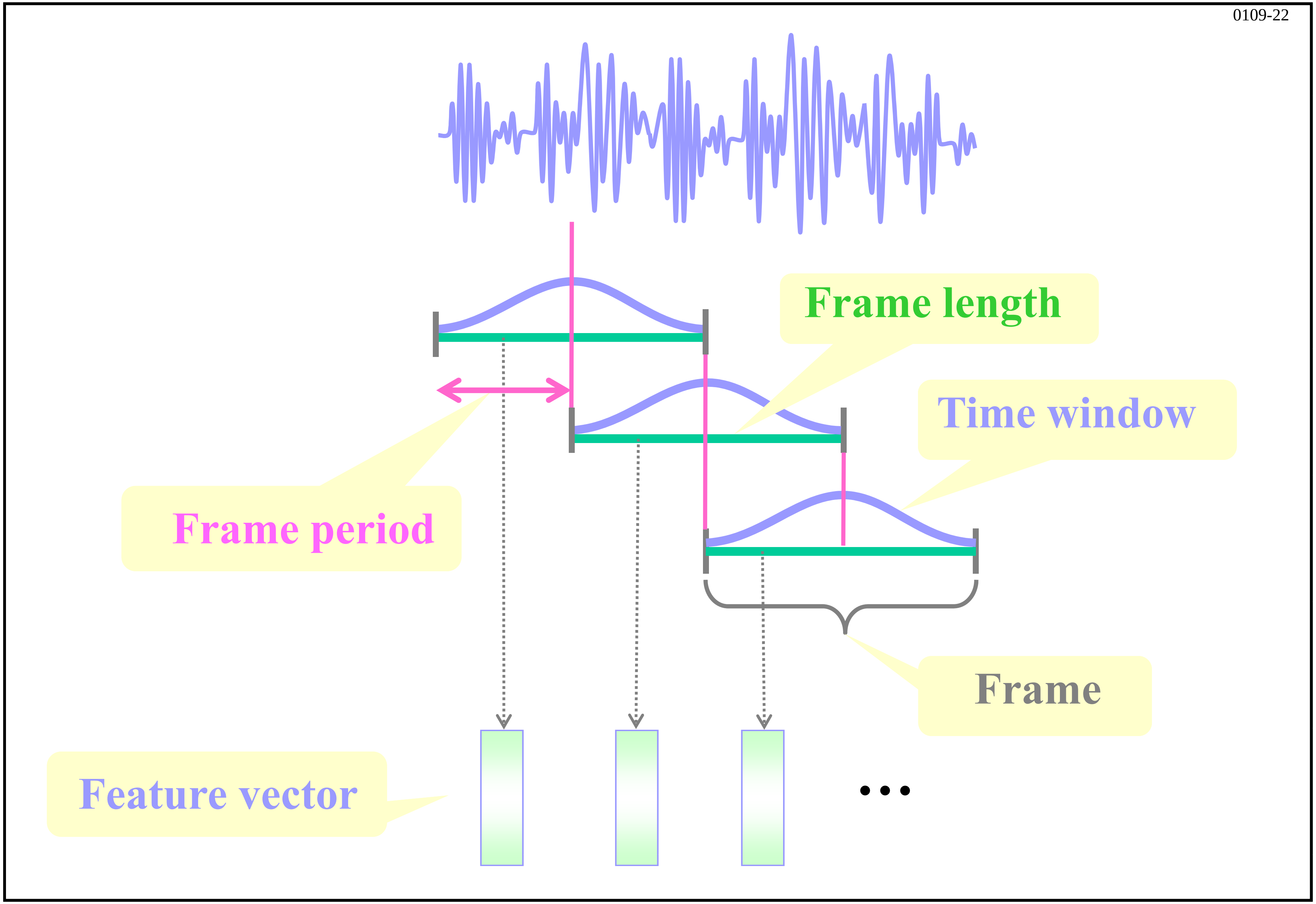


(b) Incorrect sampling ($S < 2W$)

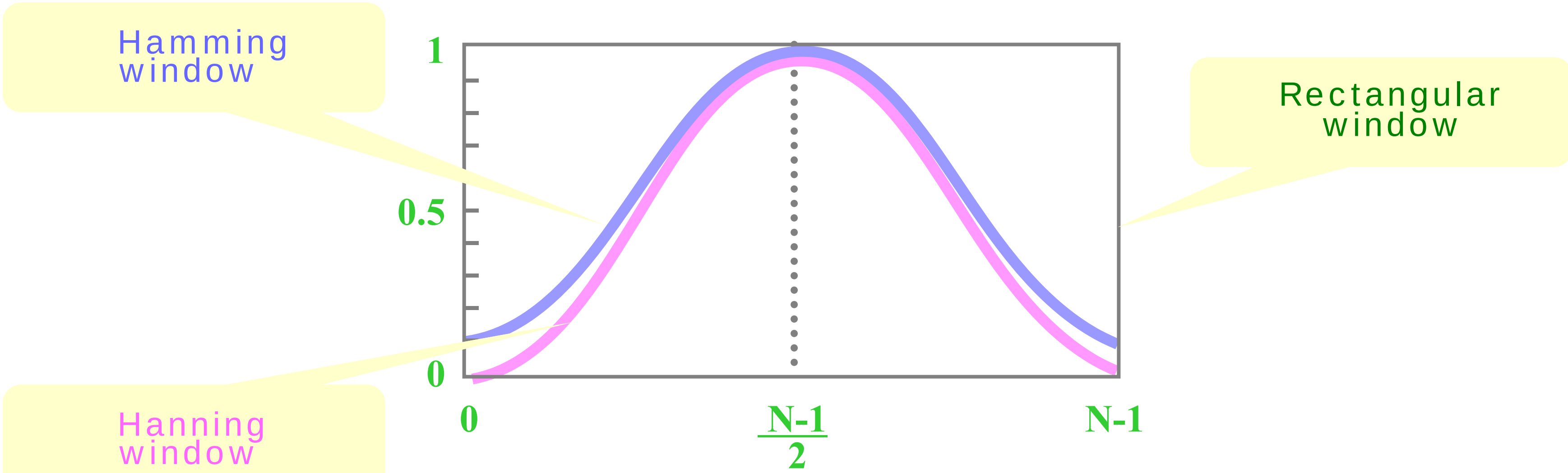
Sampling in the frequency domain



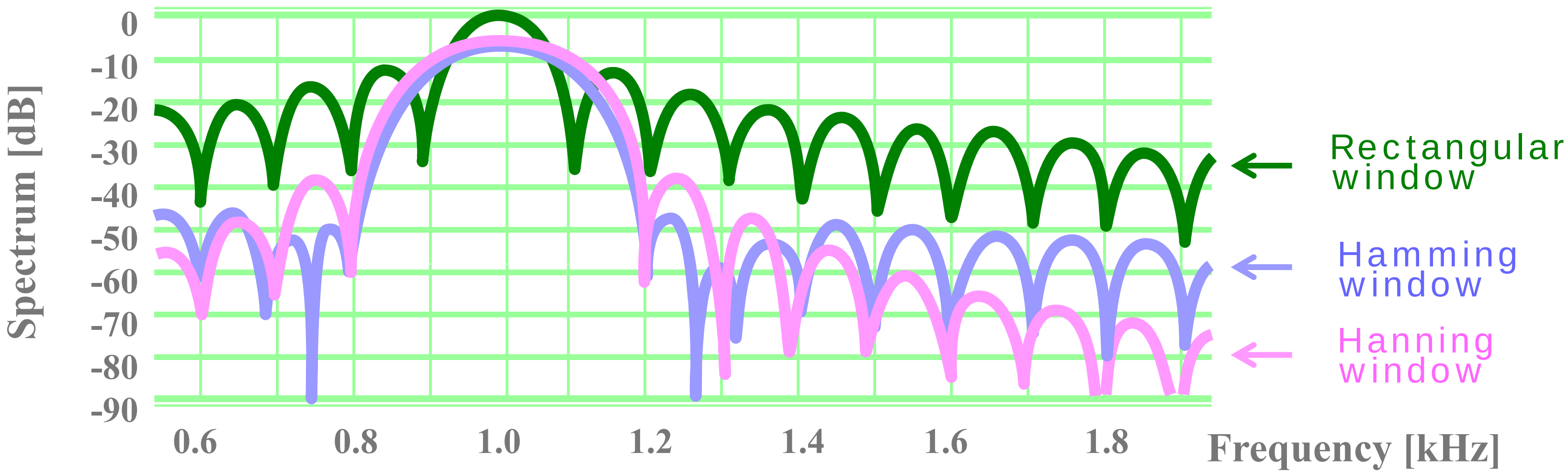
An example of the input-output characteristics of eight-level (3-bit) quantization



Feature vector (short-time spectrum) extraction from speech

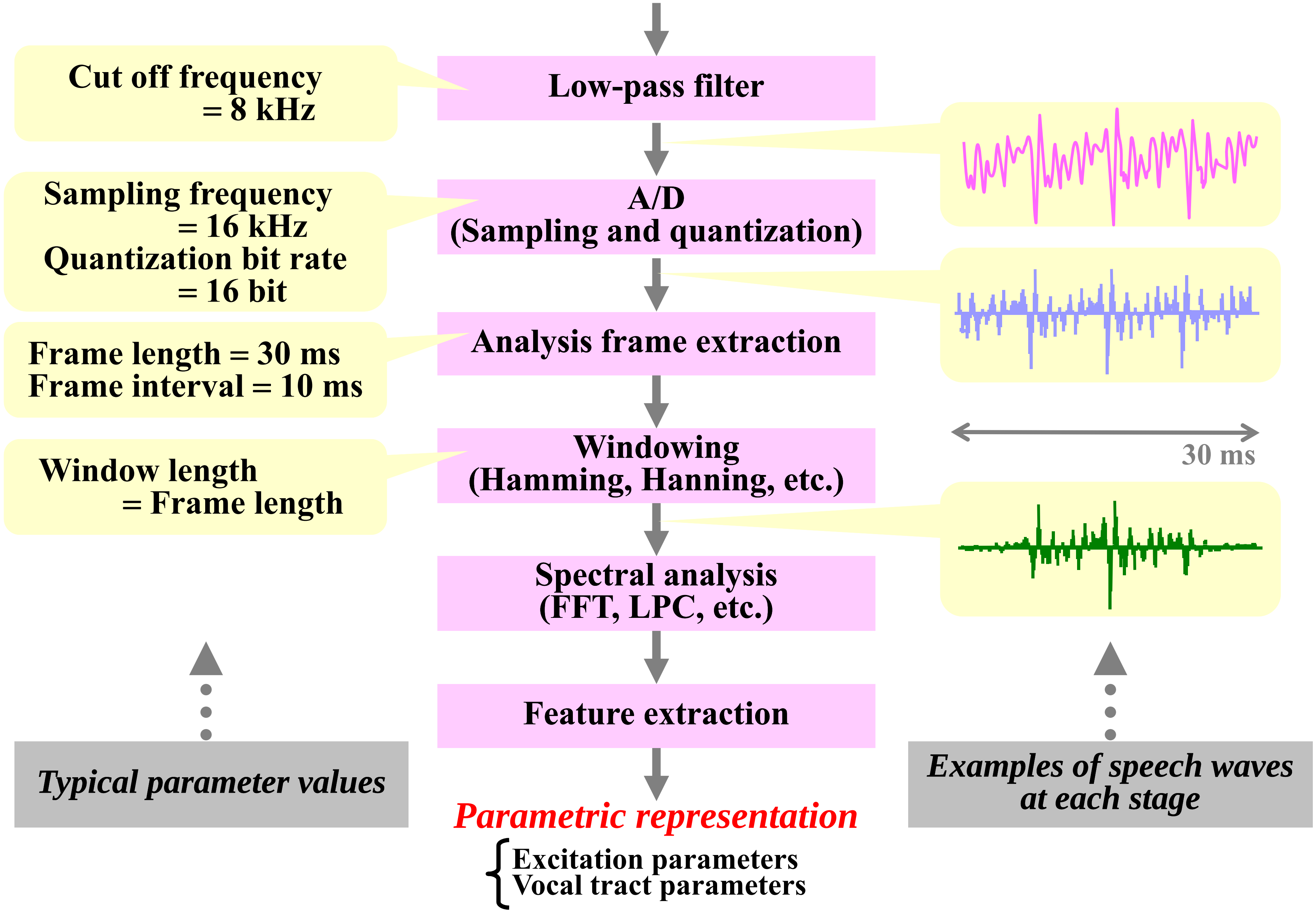


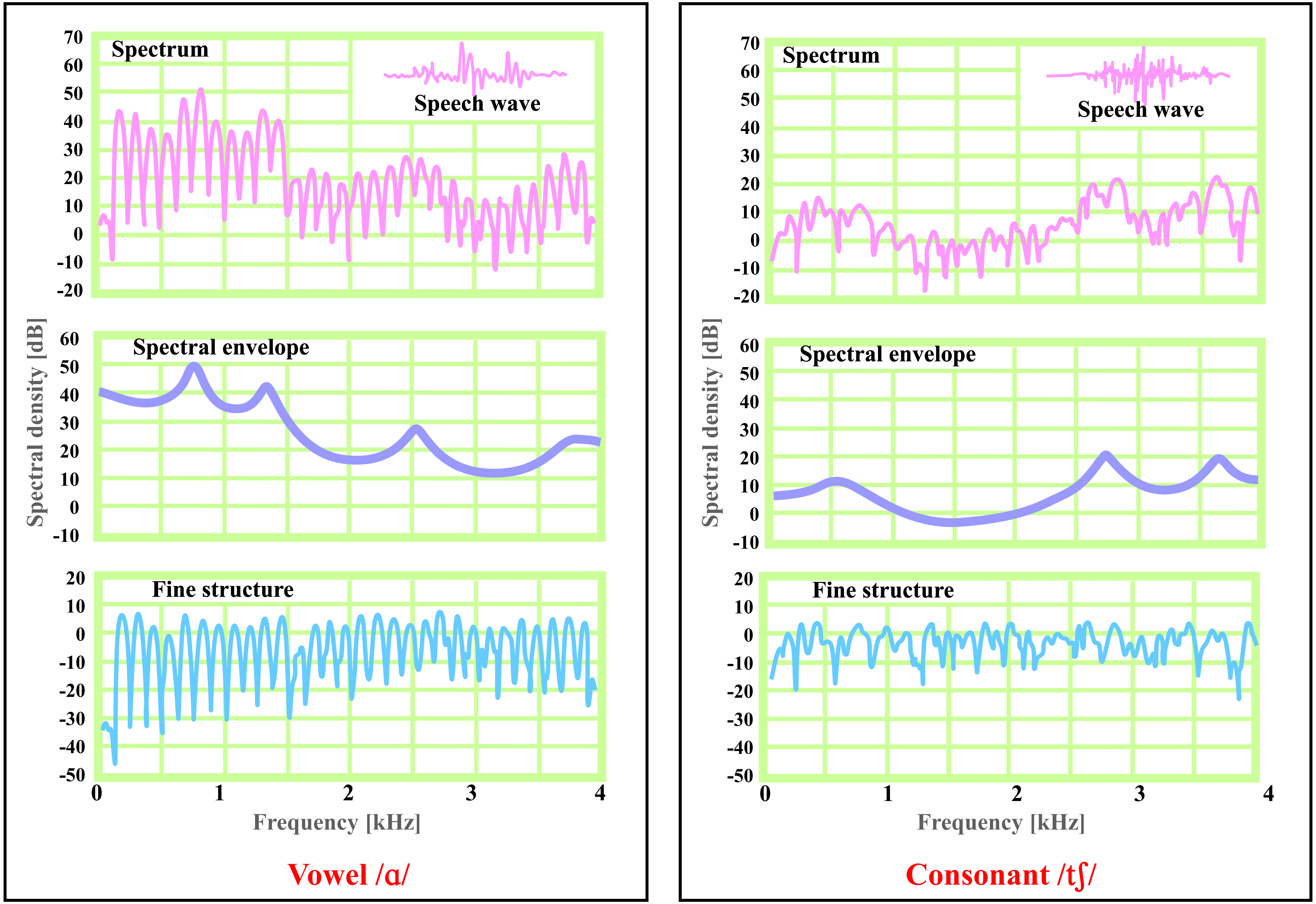
(a) Major window functions



(b) The spectrum for the 10 periods of a 1-kHz sinusoidal wave extracted using each of the windows

Block diagram of a typical speech analysis procedure.





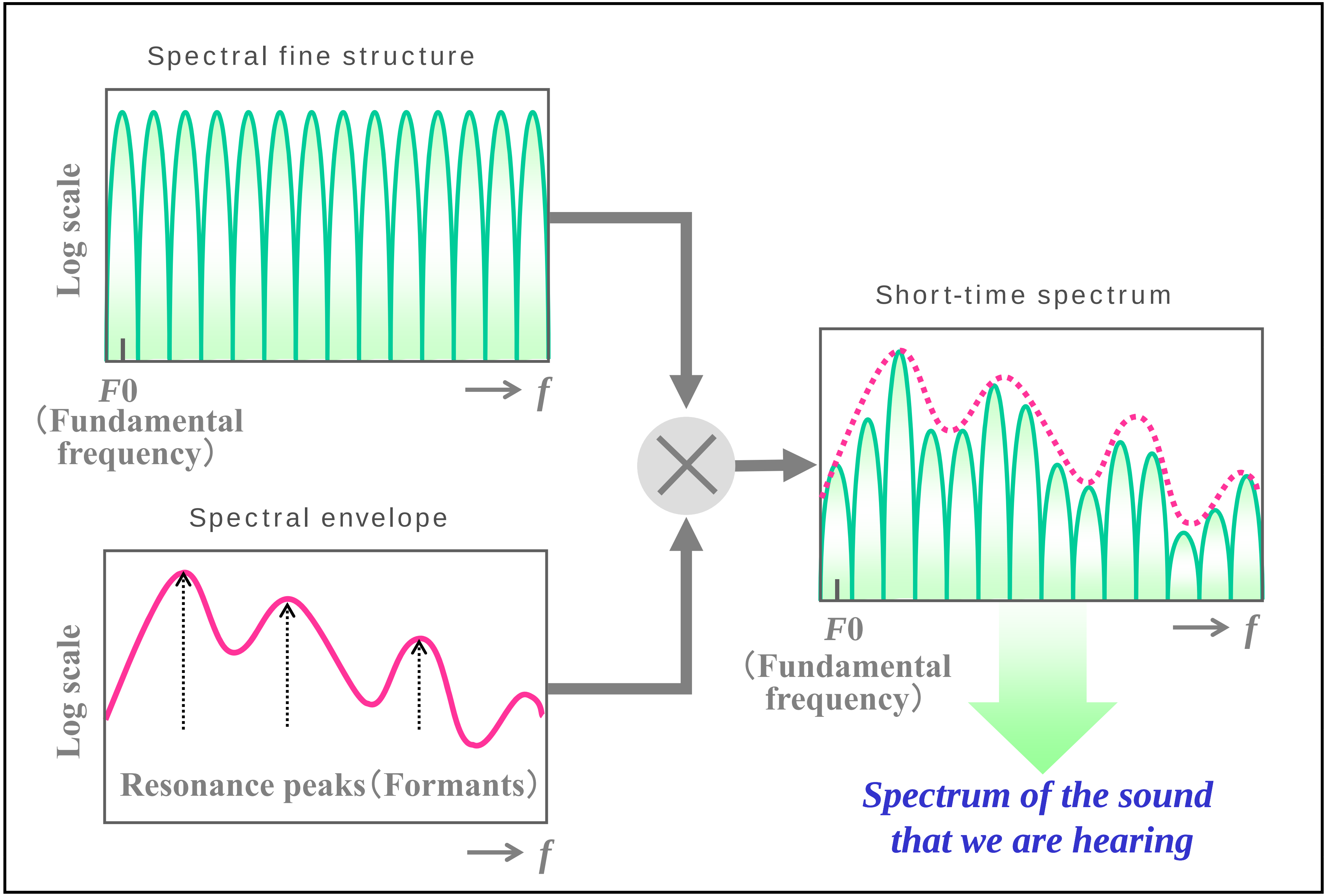
Structure of short-time speech spectra for male voices when uttering vowel /ɑ/ and consonant /tʃ/

Major methods for analyzing speech spectra and their principal features

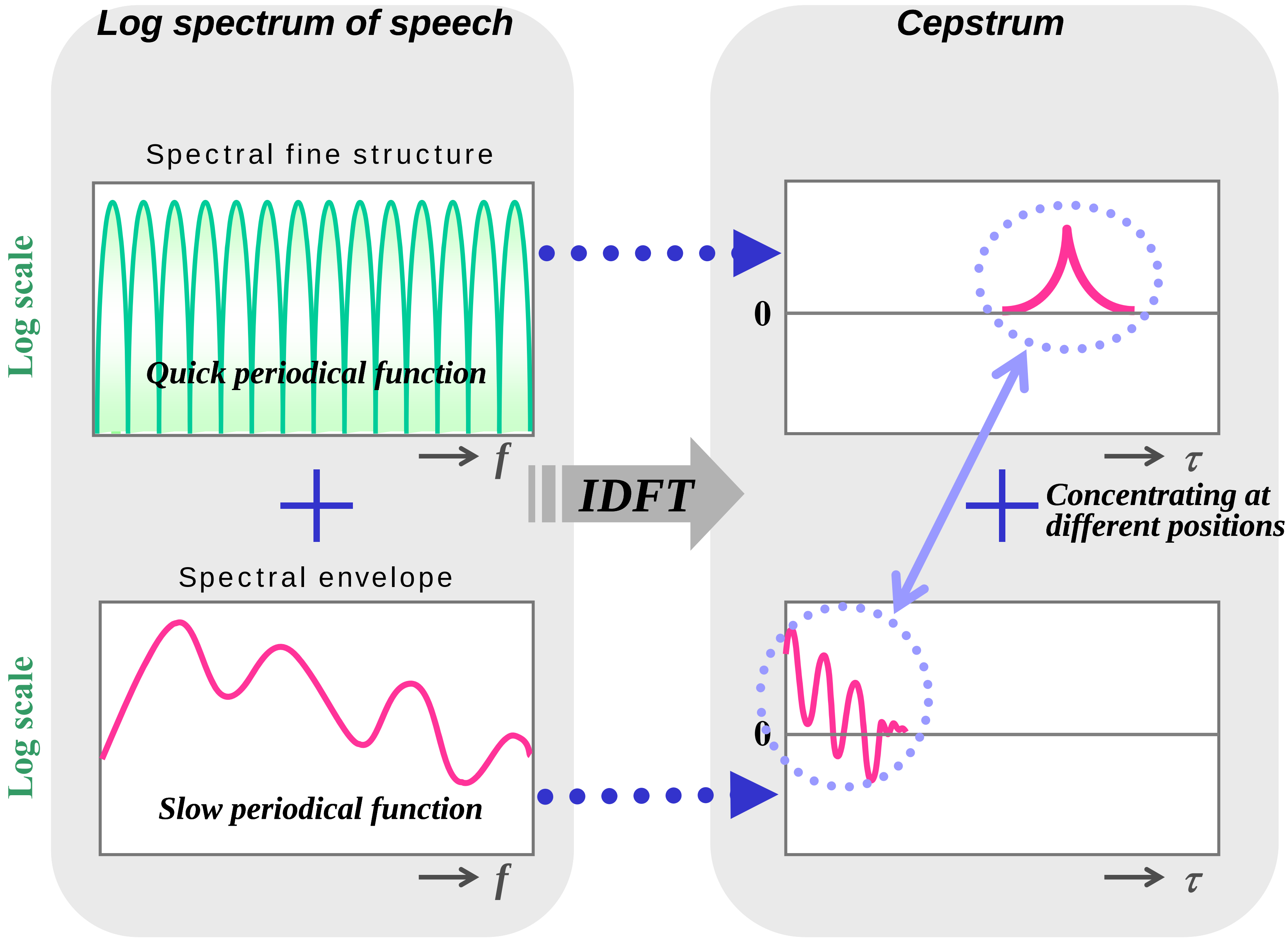
Type	Analysis methods	Parameters	Features
NPA	(i) Short-time autocorrelation	$\phi(m)$	Spectral envelope and fine structure are convoluted.
	(ii) Short-time spectrum	$S(\omega)$	Spectral envelope and fine structure are multiplied. Fast algorithm can be realized by FFT.
	(iii) Cepstrum	$c(\tau)$	Spectral envelope and fine structure can be Separated in quefrency domain. Two FFTs and log transform are necessary.
	(iv) Band-pass filter bank	rms of filter output	Global spectral envelope can be obtained.
	(v) Zero-crossing analysis	Zero-crossing rate	Formant freq. Can be obtained by Combination with (iv). Realized by simple hardware.
PA	(i) Analysis-by -synthesis	Formant, band-width, etc.	Precise modeling is possible. Accurate formant freq. can be obtained. Complicated iteration is necessary.
	(ii) Linear predictive coding		Simple all-pole spectrum modeling.Parameters can be estimated from auto-corr. or ovariance without iteration.

Major methods for analyzing speech spectra and their principal features (continued)

Type	Analysis methods	Parameters	Features
PA (cont.)	(ii-*a*) Maximum likelihood method	α_i	Stability of synthesis filter is guaranteed. Time window is necessary. Number of calculations $\propto p^2$
	(ii-*b*) Convariance method	α_i	Stability of synthesis filter is not guaranteed. Suitable for short-time analysis. Number of calculations $\propto p^3$
	(ii-*c*) PARCOR method	k_i	Normal equation can be solved by lattice filter. Equivalent to (*a*) and (*b*). Number of calculations $\propto p^2$
	(ii-*d*) LSP method	ω_i	Quantization and interpolation characteristics are good. Similar to formant. Number of calculation is slightly larger than for PARCOR.



Spectral structure of speech



Log spectrum and Cepstrum

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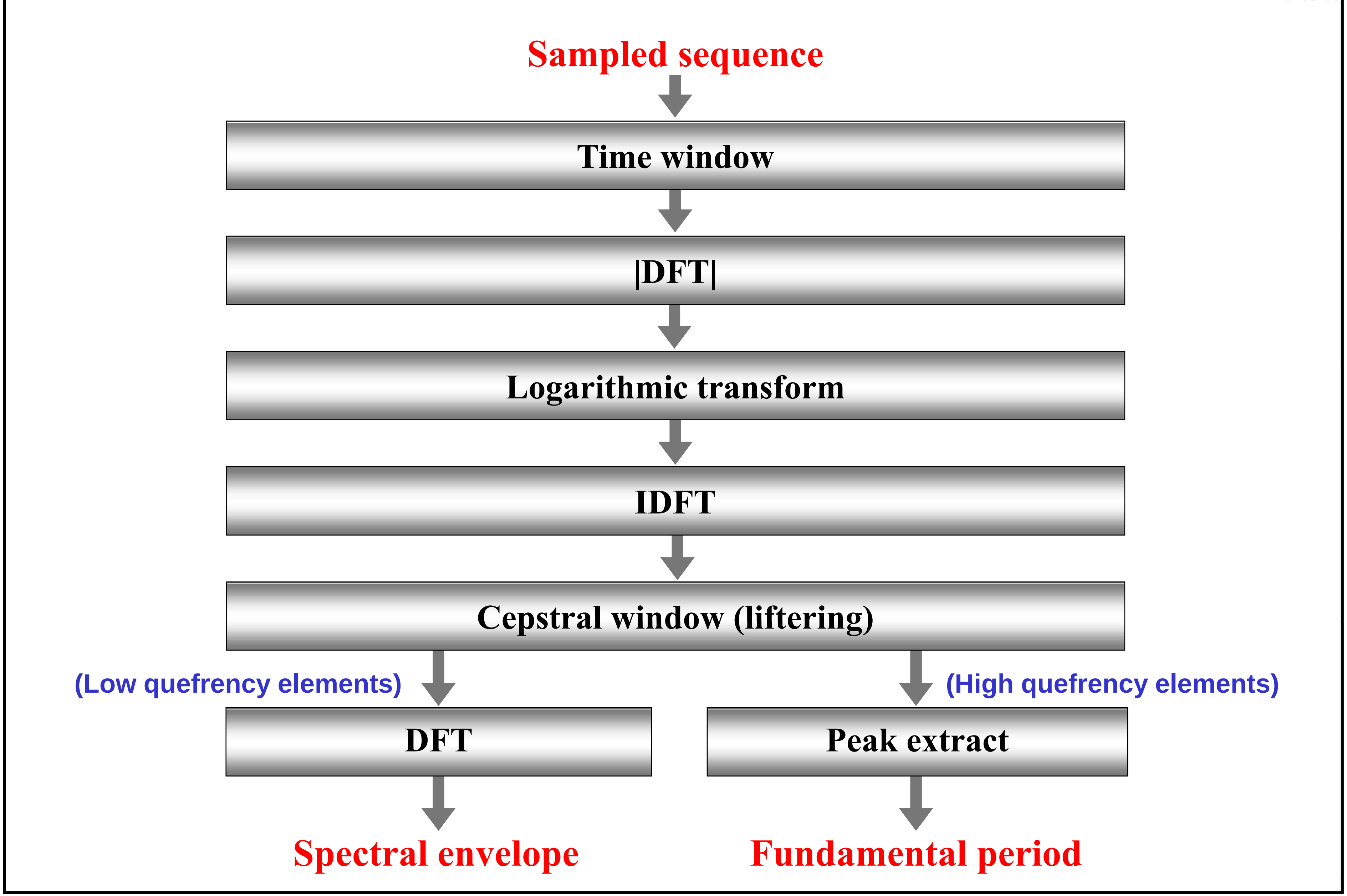
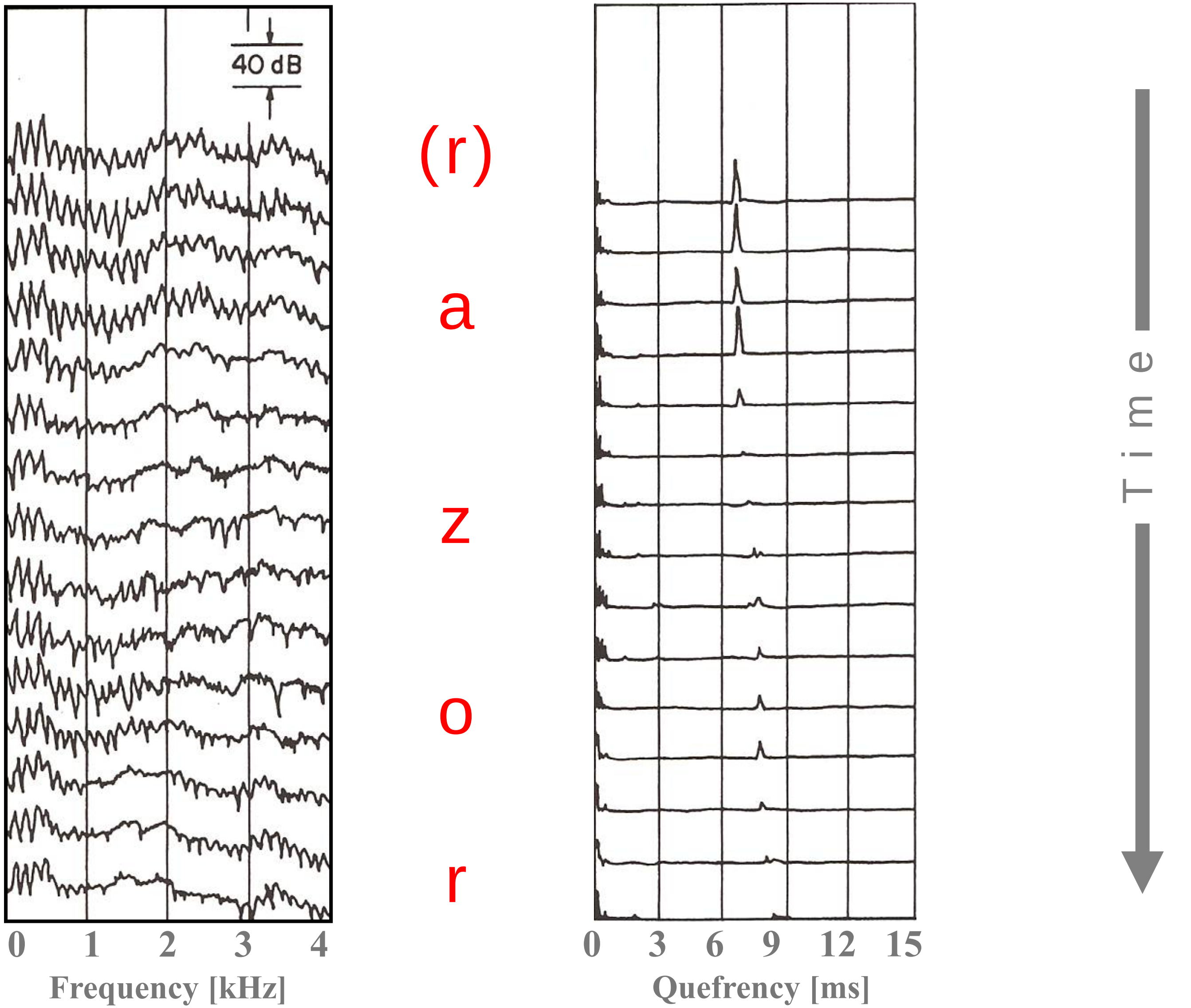


Block diagram of cepstrum analysis for extracting spectral envelope and fundamental period

Examples of short-time spectra (left) and cepstra (right) for male voice when uttering "(r)azor". Sampling frequency 10kHz; Hamming window length 40ms; frame interval 10ms.



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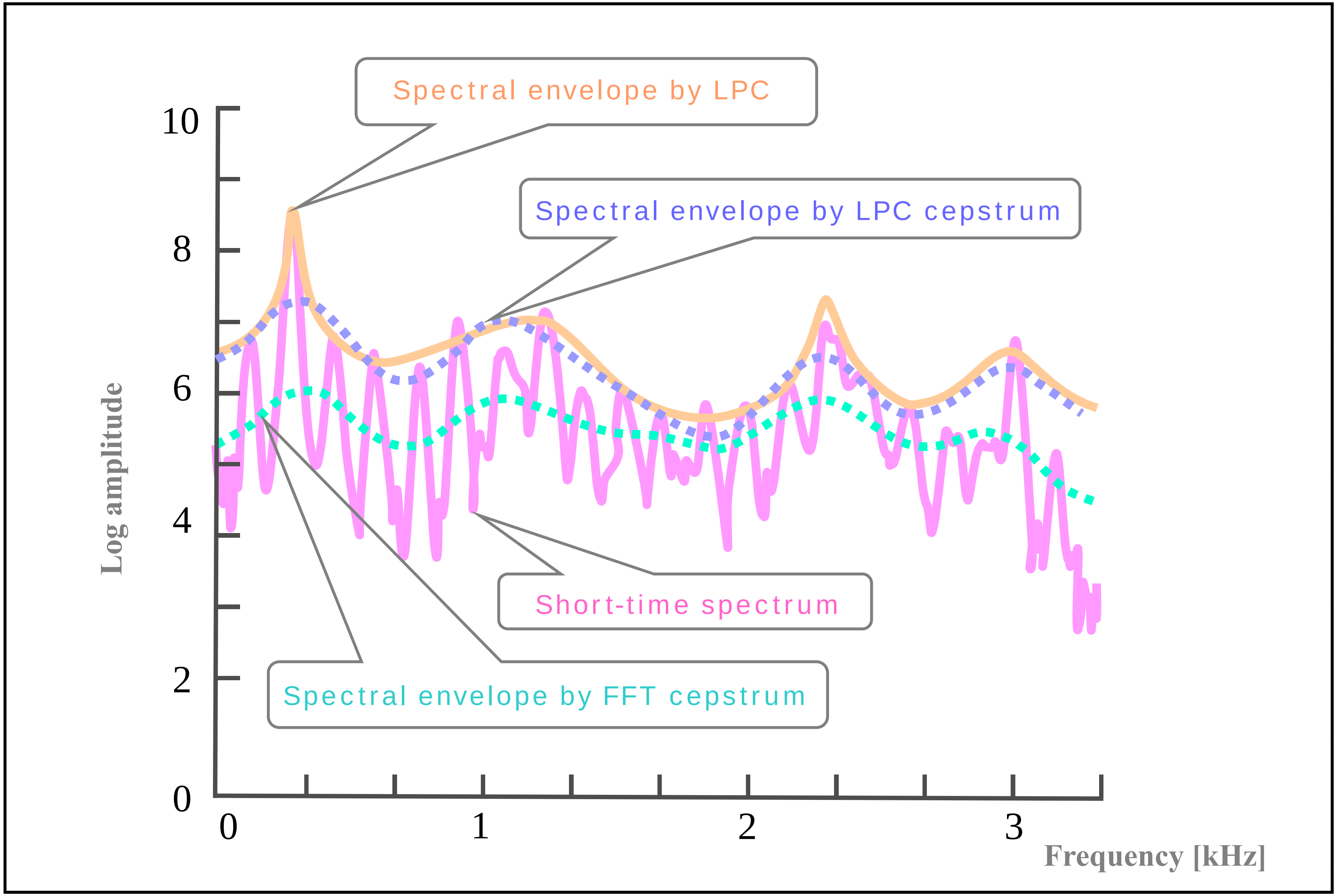
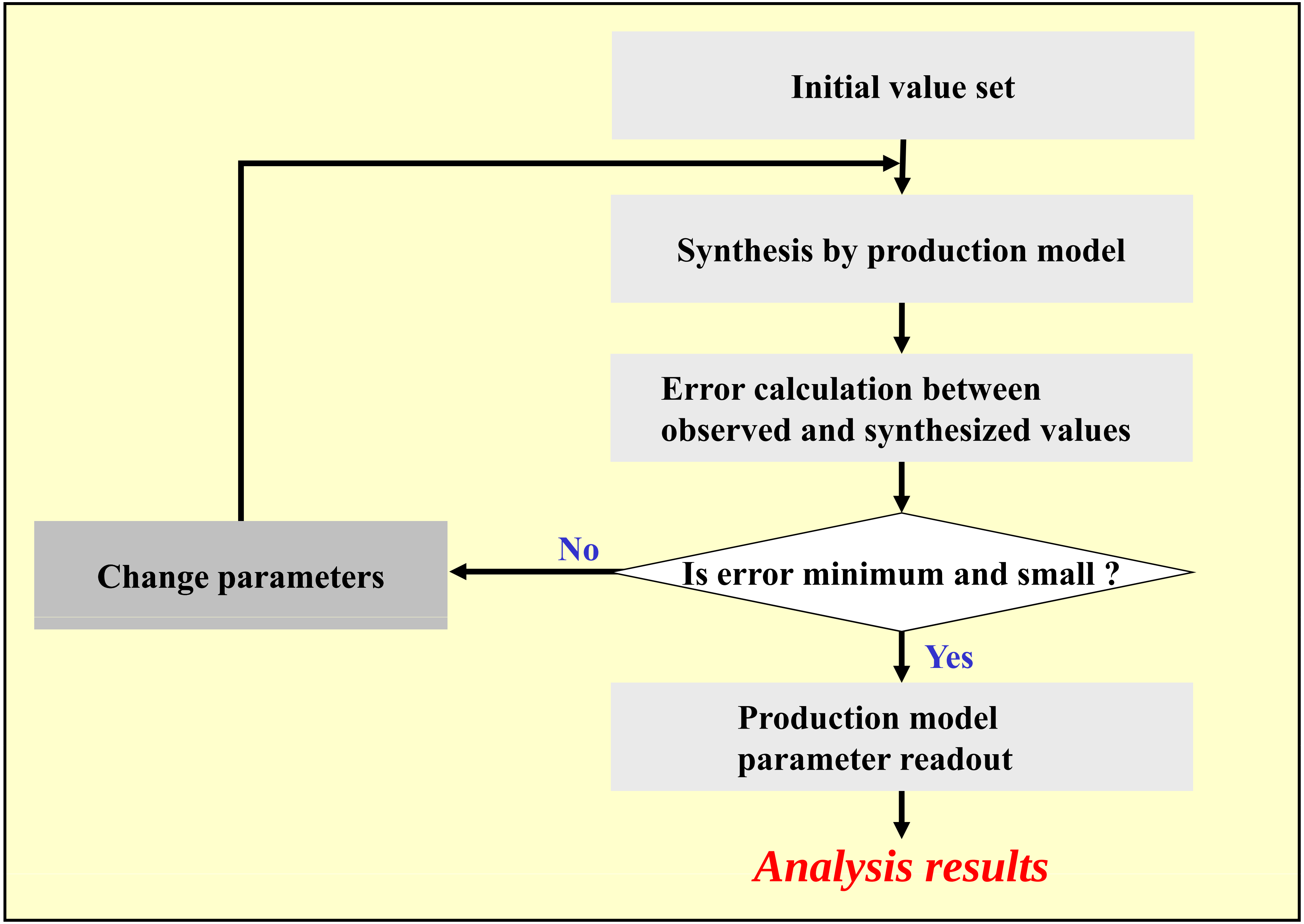


Comparison of spectral envelopes by LPC, LPC cepstrum, and FFT cepstrum methods



Principle of analysis-by-synthesis (A-b-S) method

Linear prediction

(Time-domain representation)

$$x_t + \underbrace{\sum_{i=1}^{p} \alpha_i x_{t-i}}_{-\hat{x}_t} = \varepsilon_t$$

$$x_t - \hat{x}_t = \varepsilon_t$$

All-pole/Auto-regression model

(Spectral-domain representation)

$$H(z) = \frac{1}{1+\sum_{i=1}^{p} \alpha_i z^{-i}} = \frac{1}{\prod_{i=1}^{p} (1 - \frac{z}{z_i})}$$

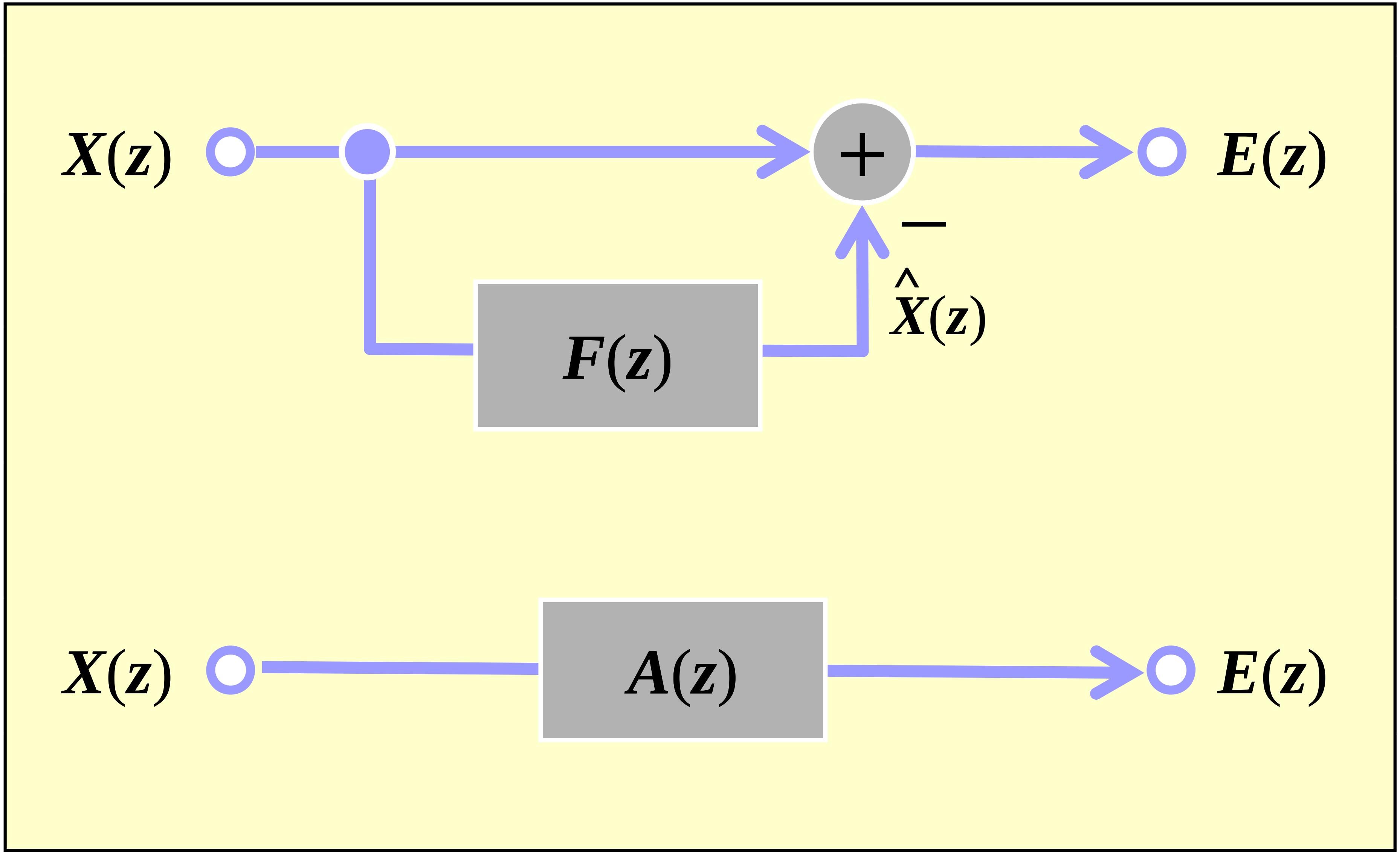
LMS estimation

Maximum-likelihood estimation

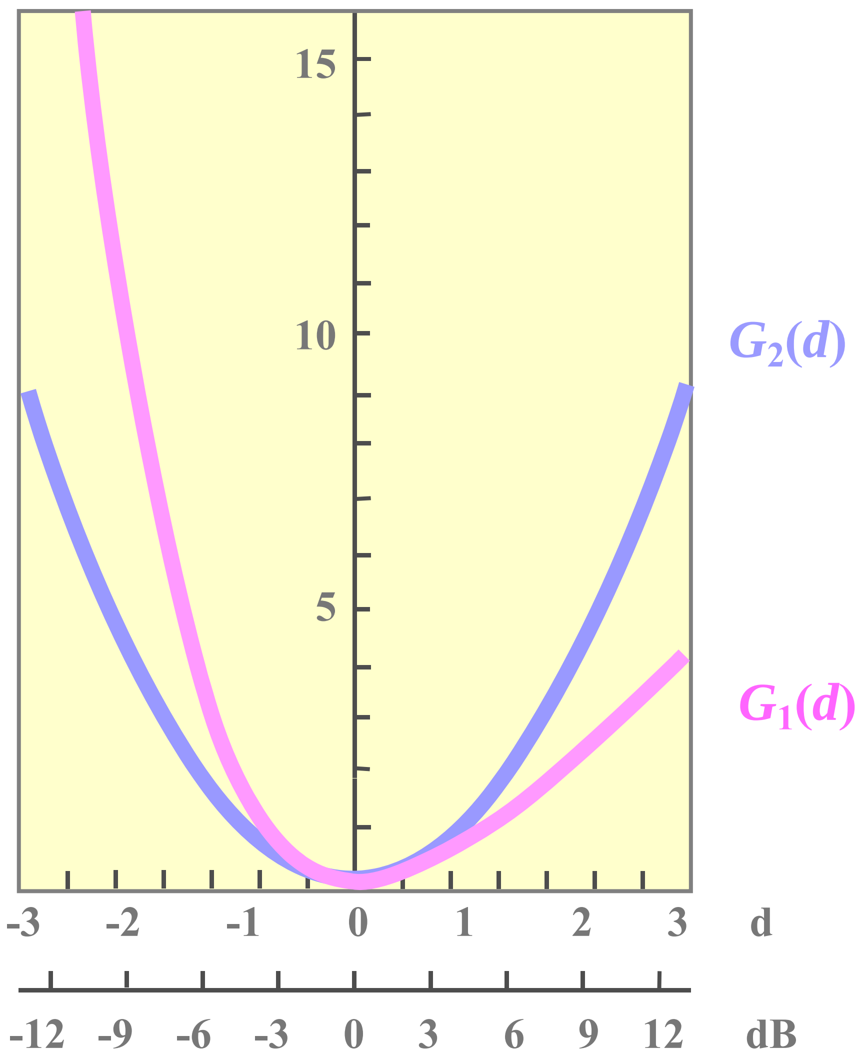
Normal equation

$$\begin{pmatrix} r(0) & r(1) & \cdots & r(p-1) \\ r(1) & r(0) & & \vdots \\ \vdots & & r(0) & r(1) \\ r(p-1) & \cdots & r(1) & r(0) \end{pmatrix} \begin{pmatrix} \alpha_1 \\ \alpha_2 \\ \vdots \\ \alpha_p \end{pmatrix} = - \begin{pmatrix} r(1) \\ r(2) \\ \vdots \\ r(p) \end{pmatrix}$$

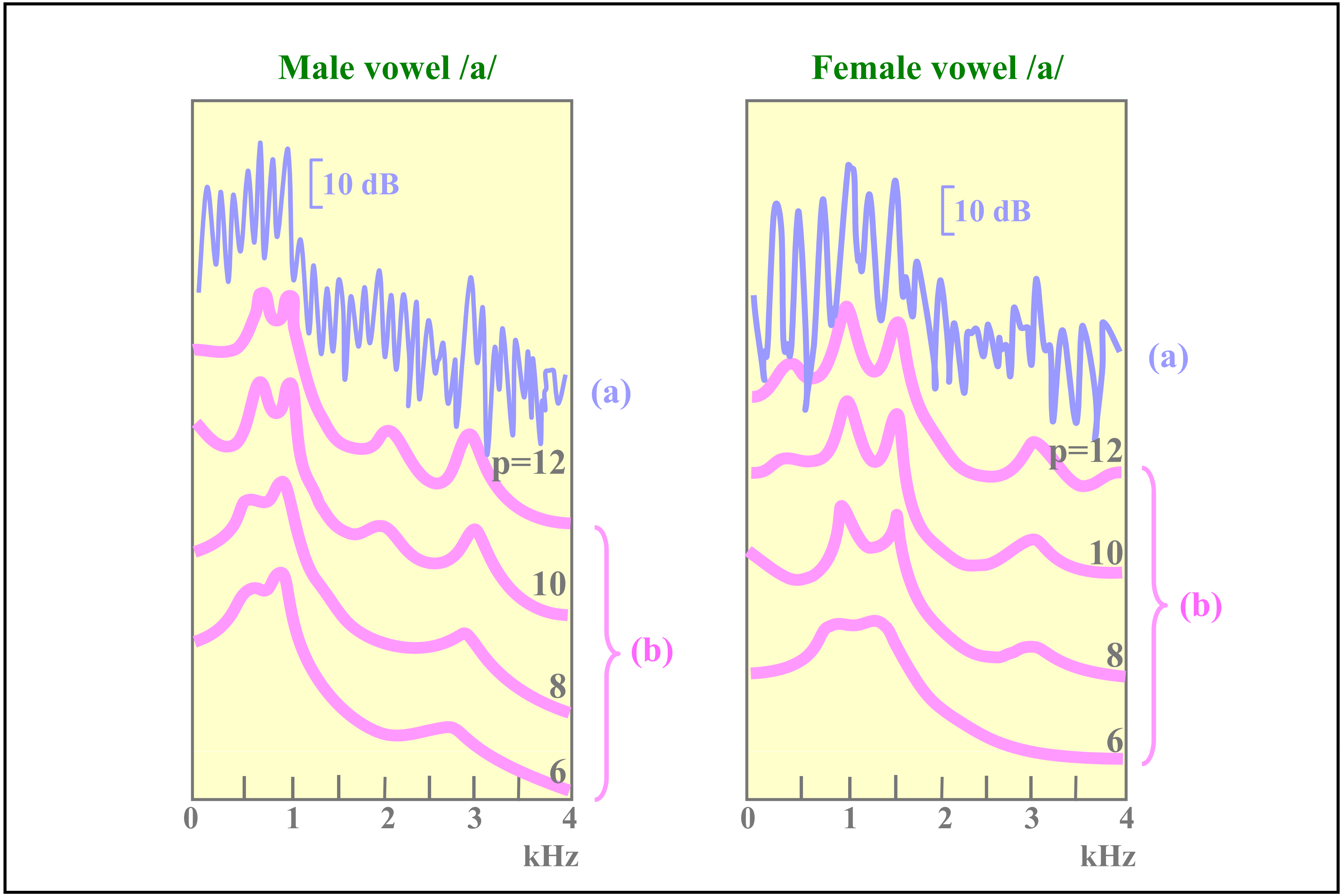
Principles of LPC analysis



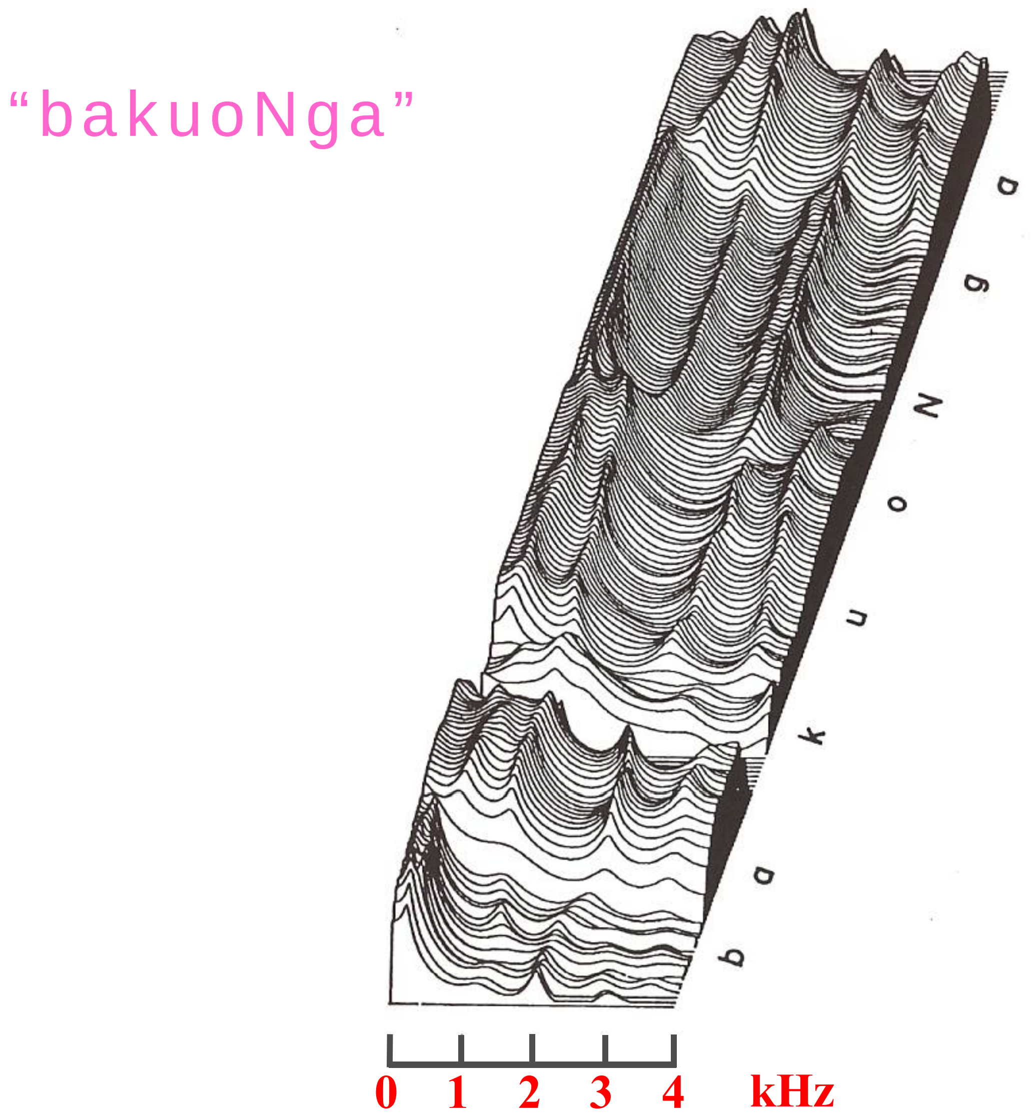
Linear prediction model block diagram



Comparison of matching error measure in maximum likelihood method, $G_1(d)$, with that in analysis-by-synthesis (A-b-S) method, $G_2(d)$.
$d = \log\{f(\lambda) / \hat{f}(\lambda)\}$; $f(\lambda)$ = model spectrum; $\hat{f}(\lambda)$ = short-term spectrum.

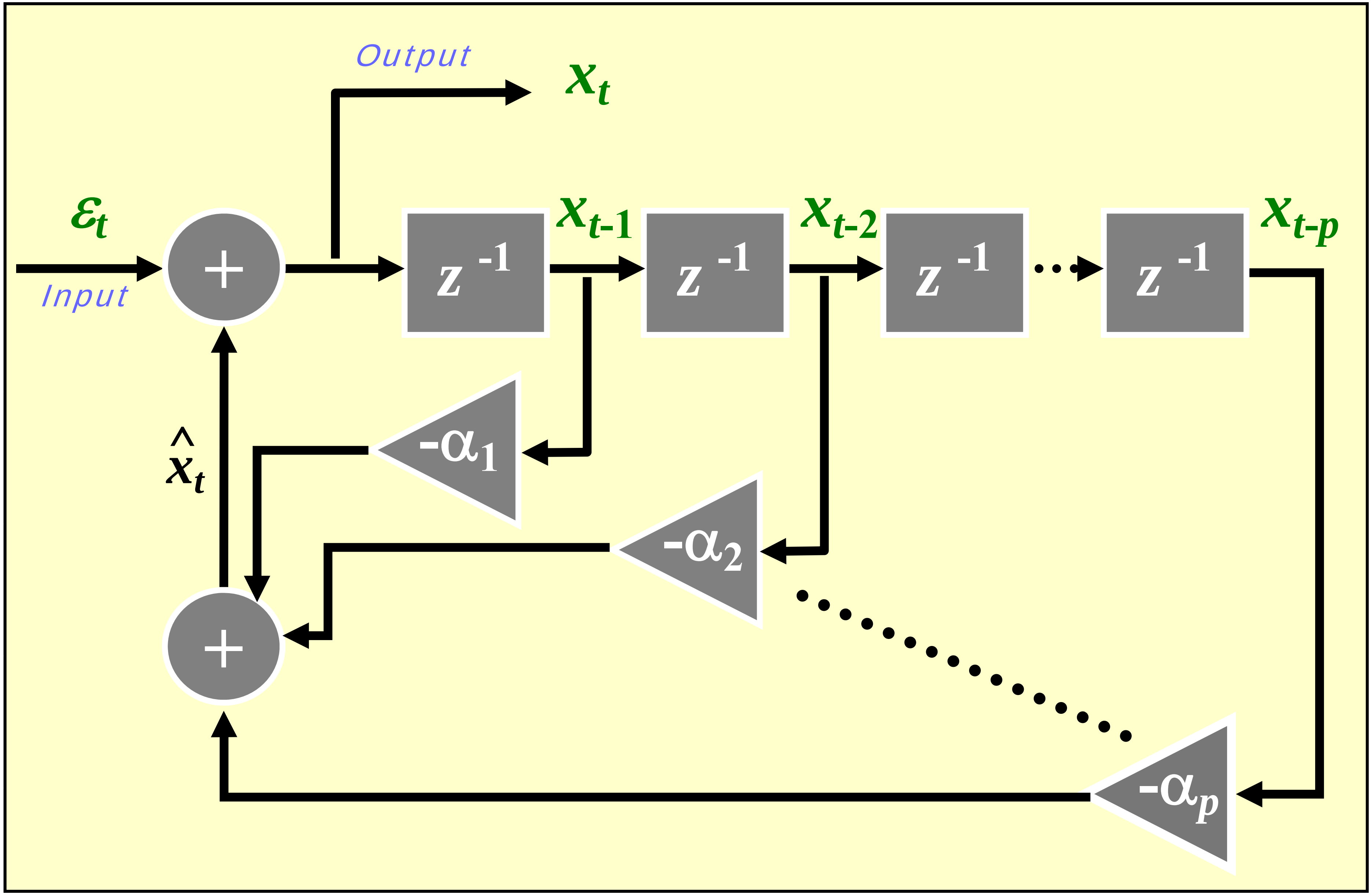


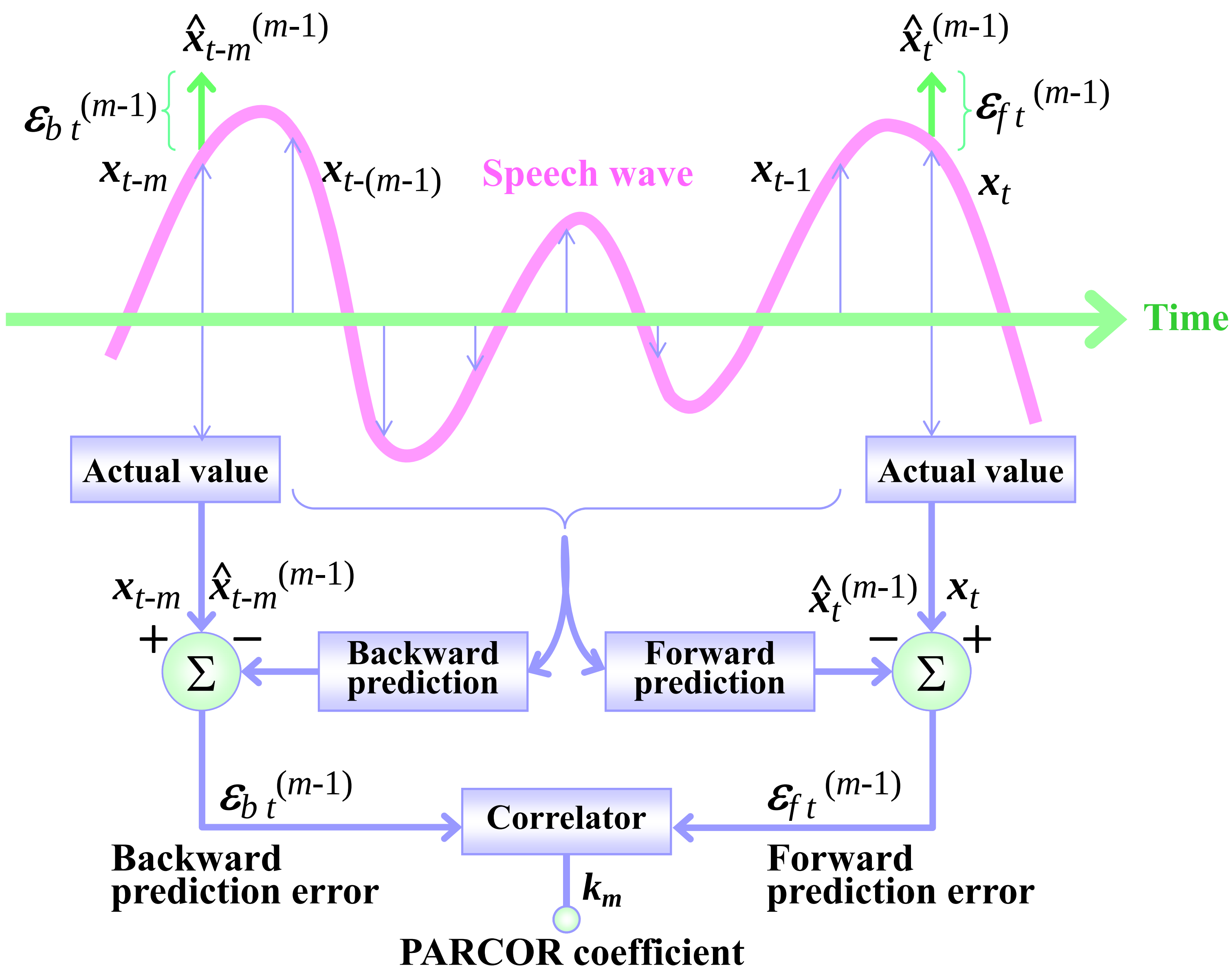
Comparison of (a) short-term spectra and (b) spectral envelopes obtained by the maximum likelihood method



Time function of spectral envelopes for the Japanese phrase /bakuoNga/ uttered by a male speaker

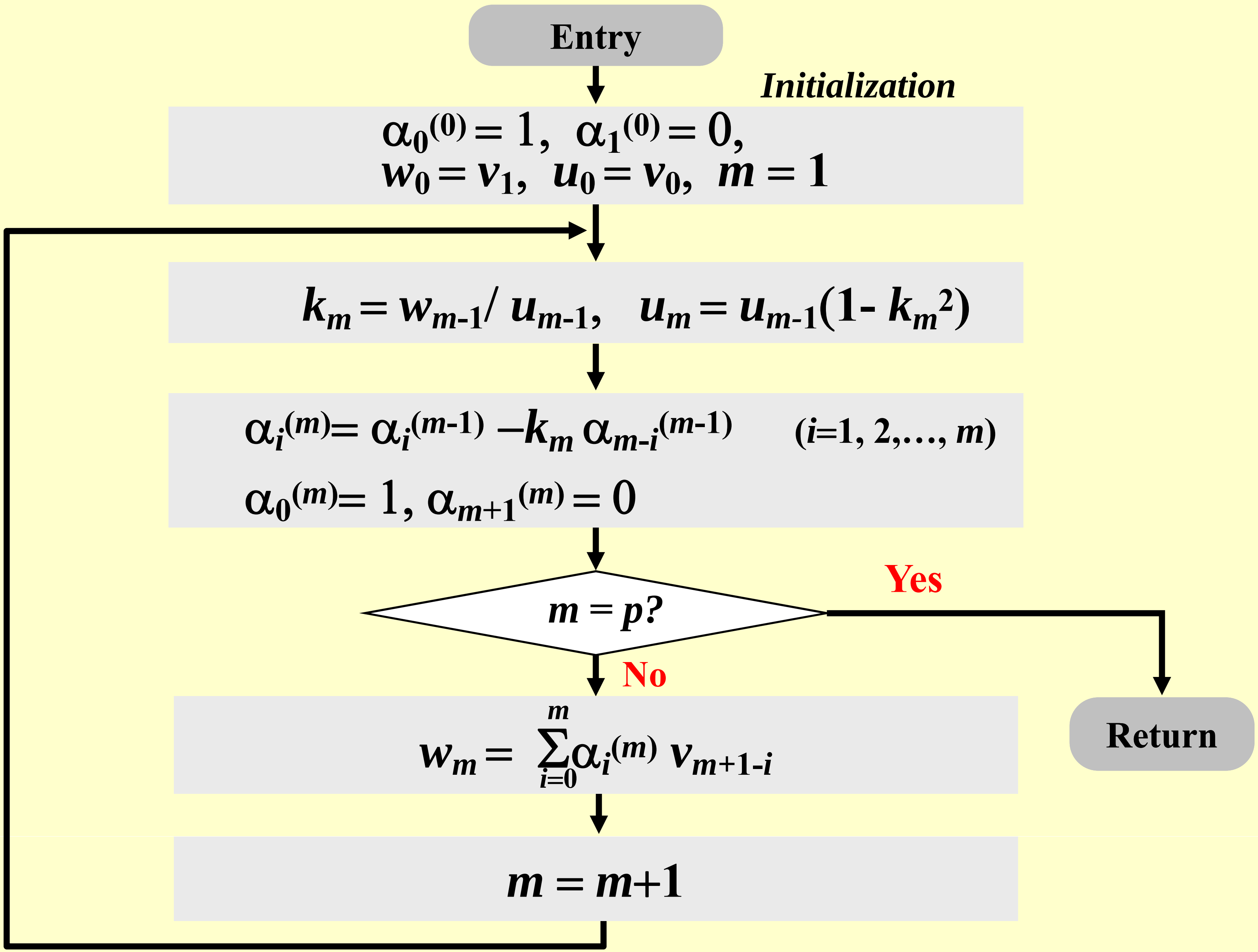
Speech synthesis circuit based on linear predictive analysis method

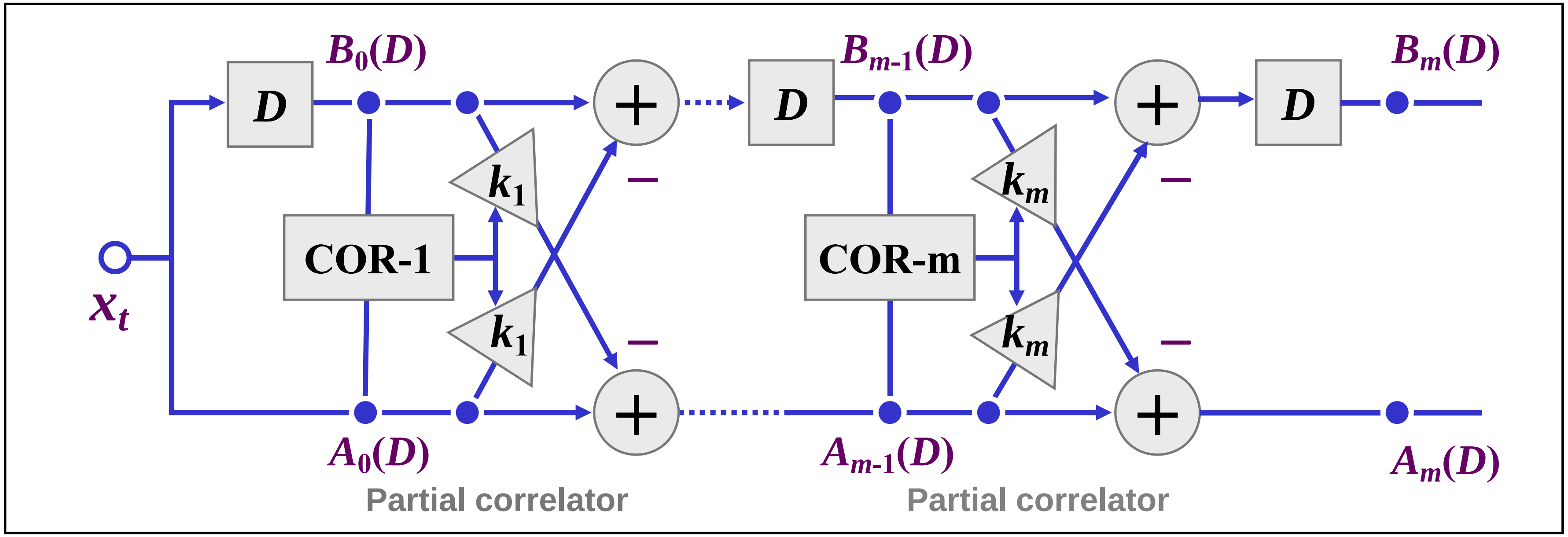




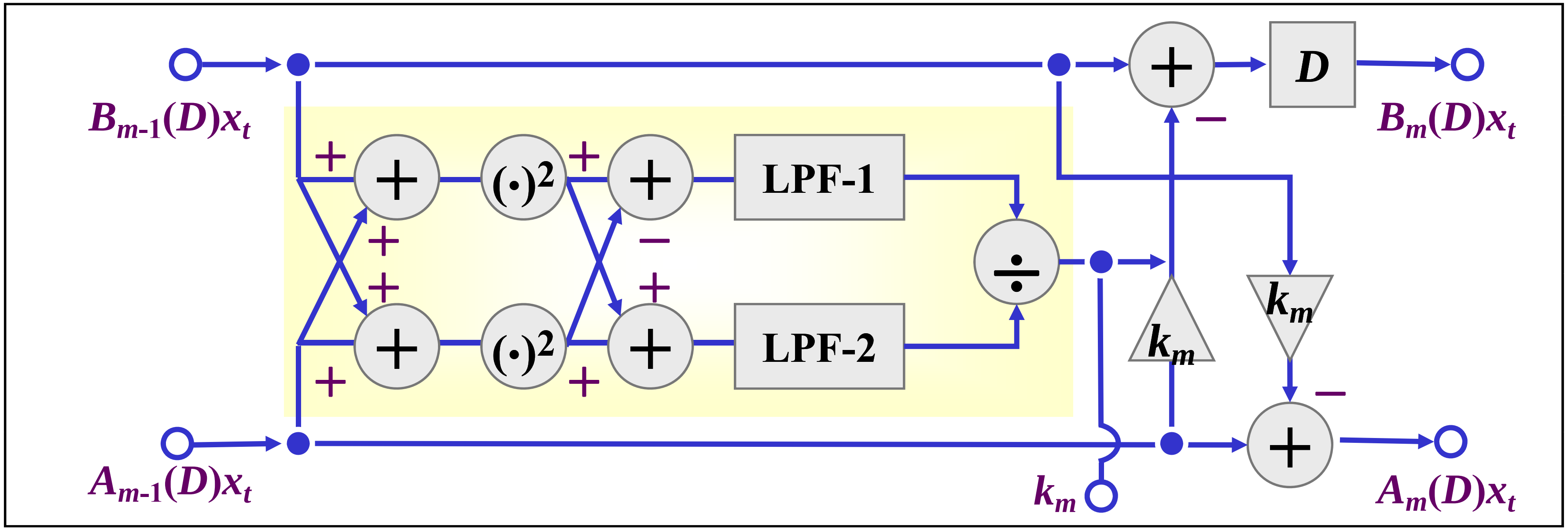
Definition of PARCOR coefficients

Flowchart for calculating $\{k_m\}_{i=0}^{p}$ and $\{\alpha_i\}_{i=0}^{p}$ from $\{v_i\}_{i=0}^{p}$

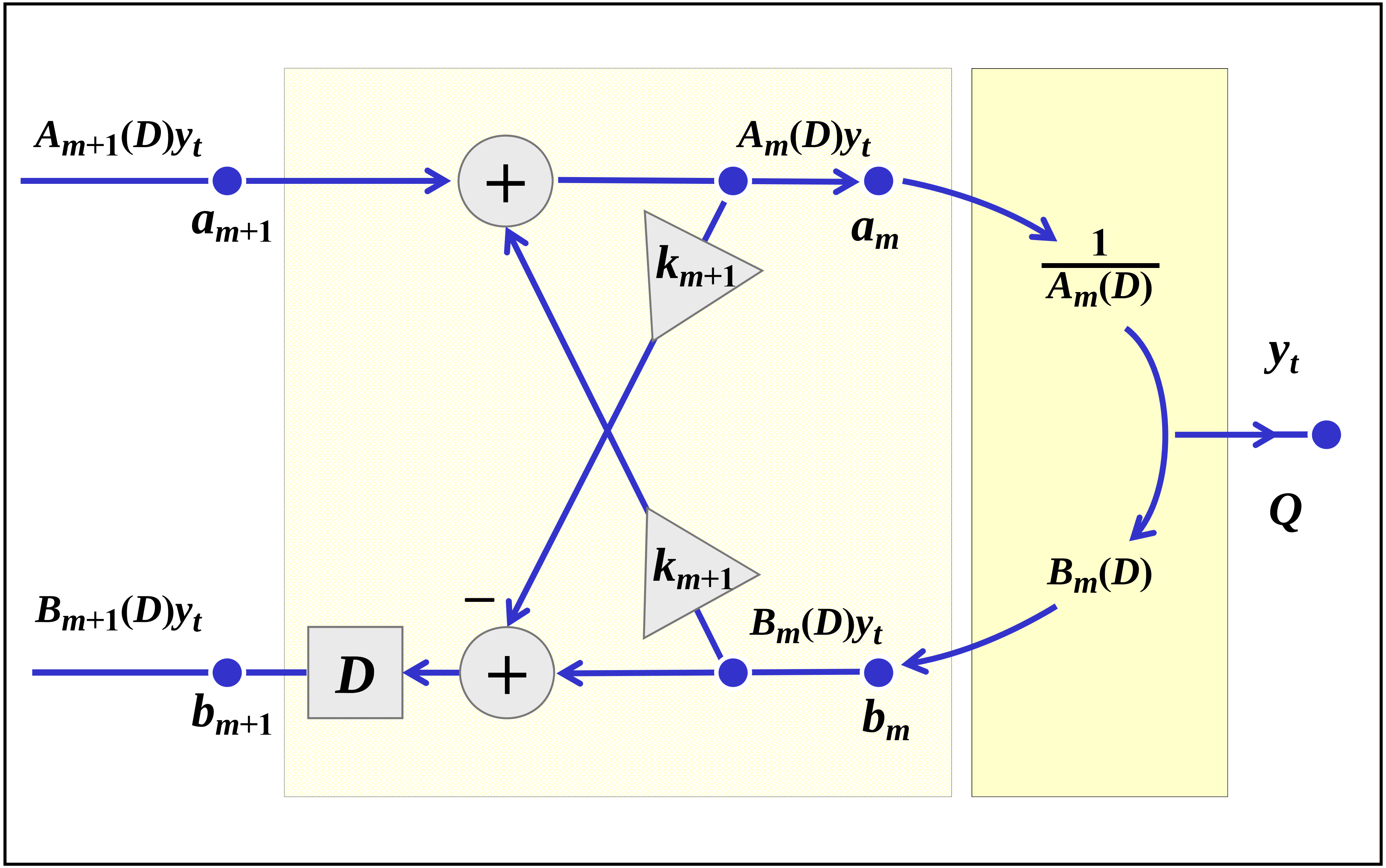




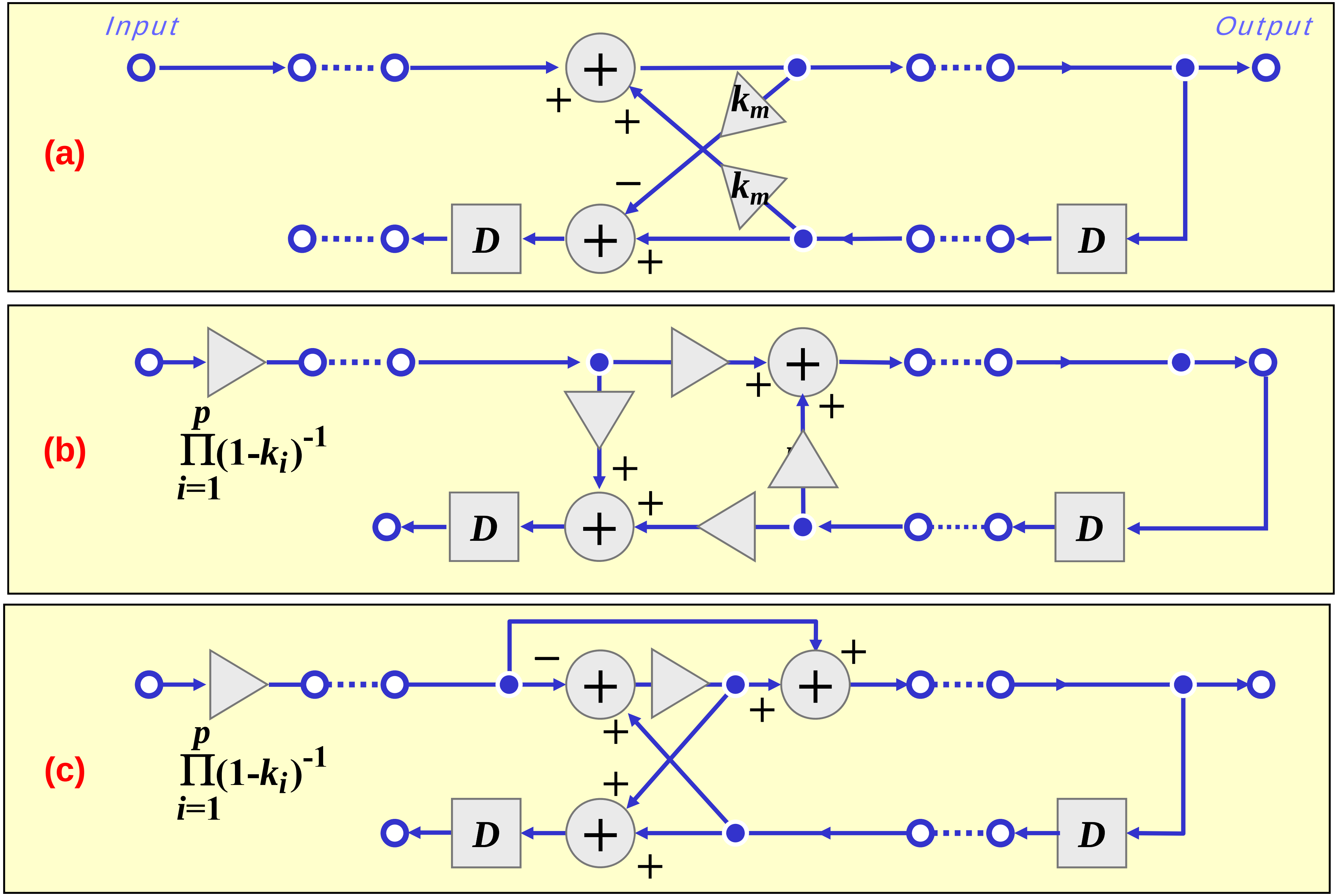
(a) PARCOR coefficient extraction circuit constructed by cascade connection of partial autocorrelators



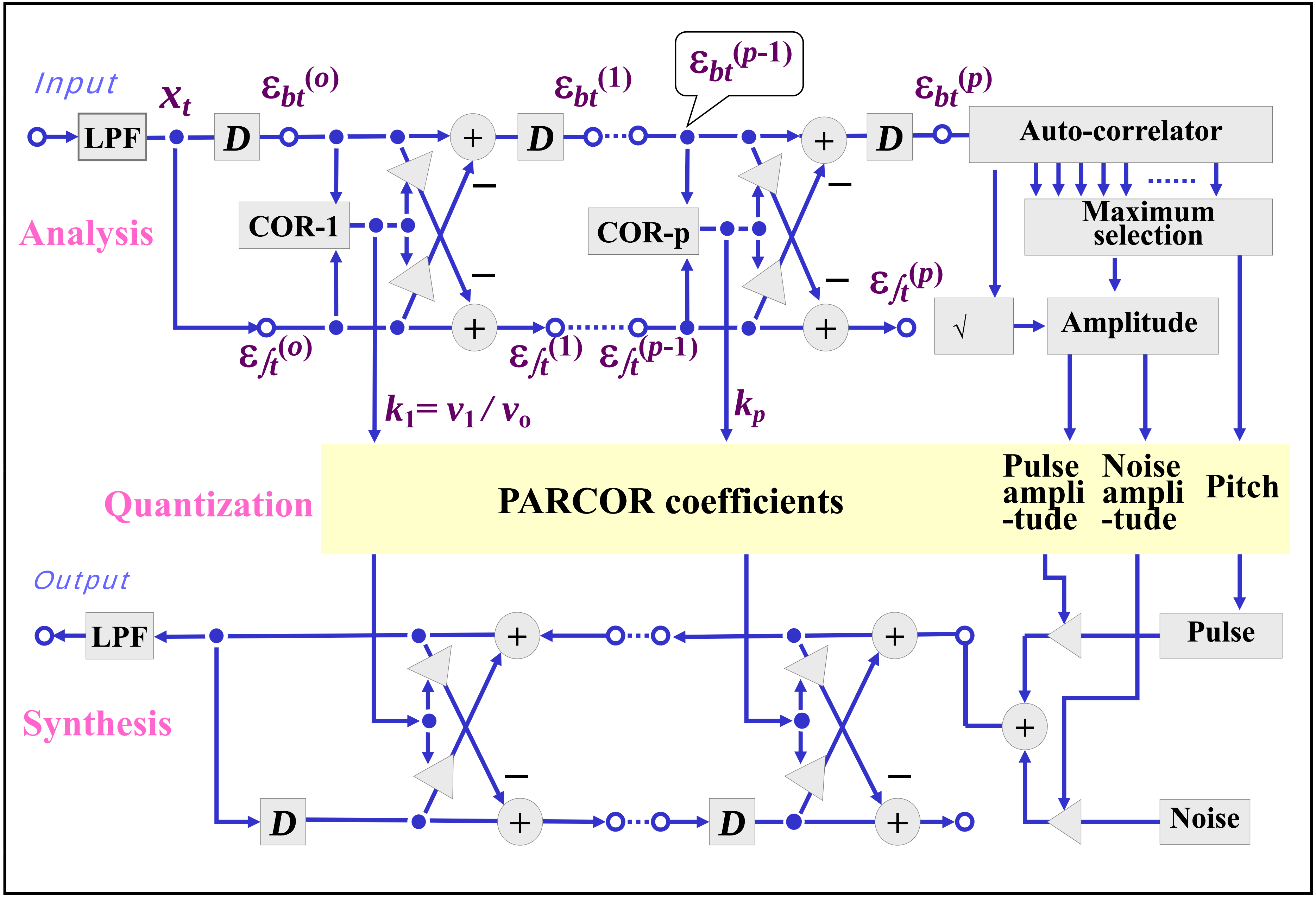
(b) Construction of each partial autocorrelator



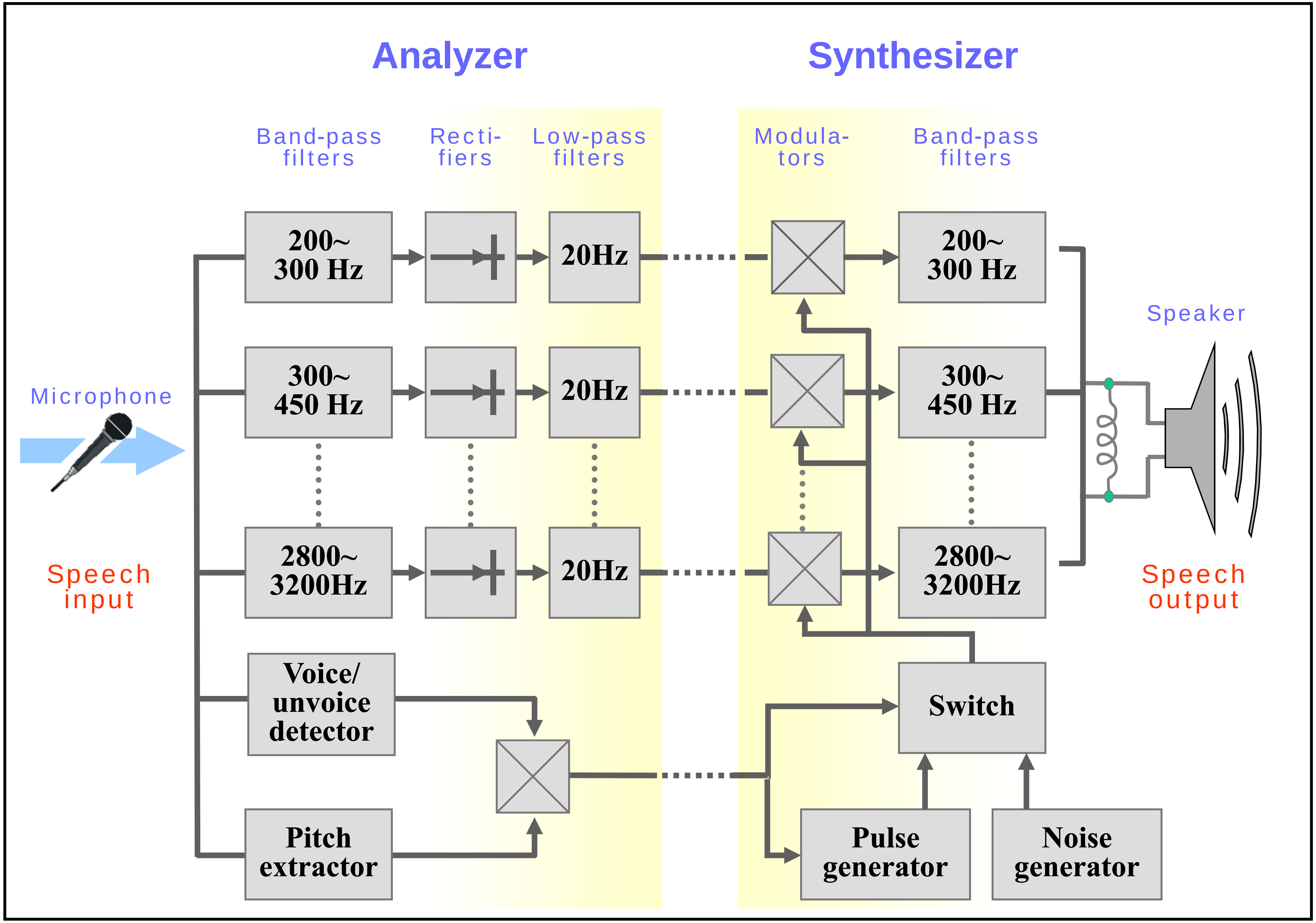
Principal construction features of synthesis filter using PARCOR coefficients



Equivalent transformations for lattice-type digital filter

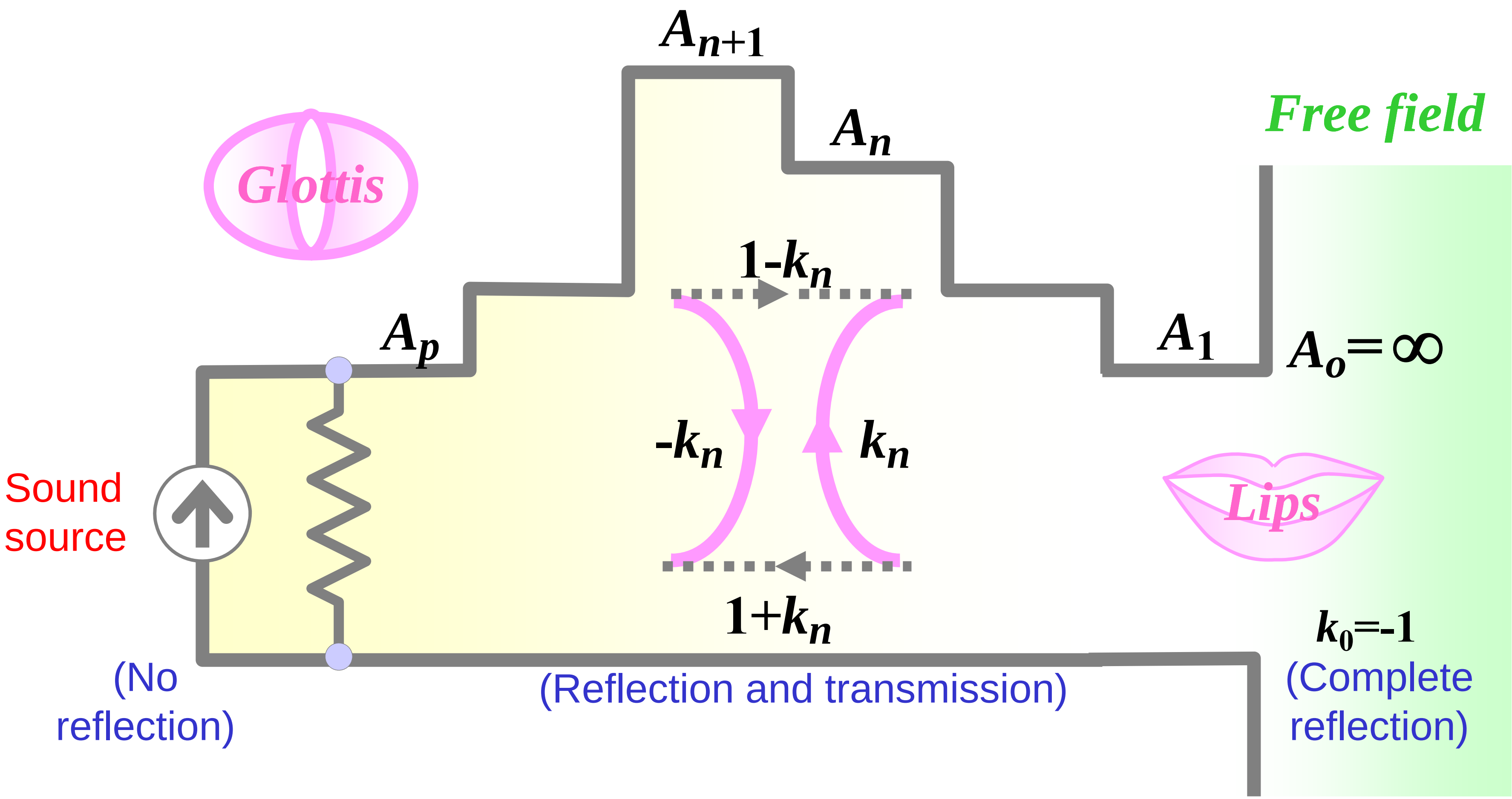


Structure of PARCOR analysis-synthesis system



Structure of the (channel) vocoder

Relationship between PARCOR coefficients $\{k_n\}$ and vocal tract area function $\{A_n\}$



$$k_n = \frac{A_{n+1} - A_n}{A_{n+1} + A_n}$$

Examples of spectral envelopes and estimated area function for five vowels

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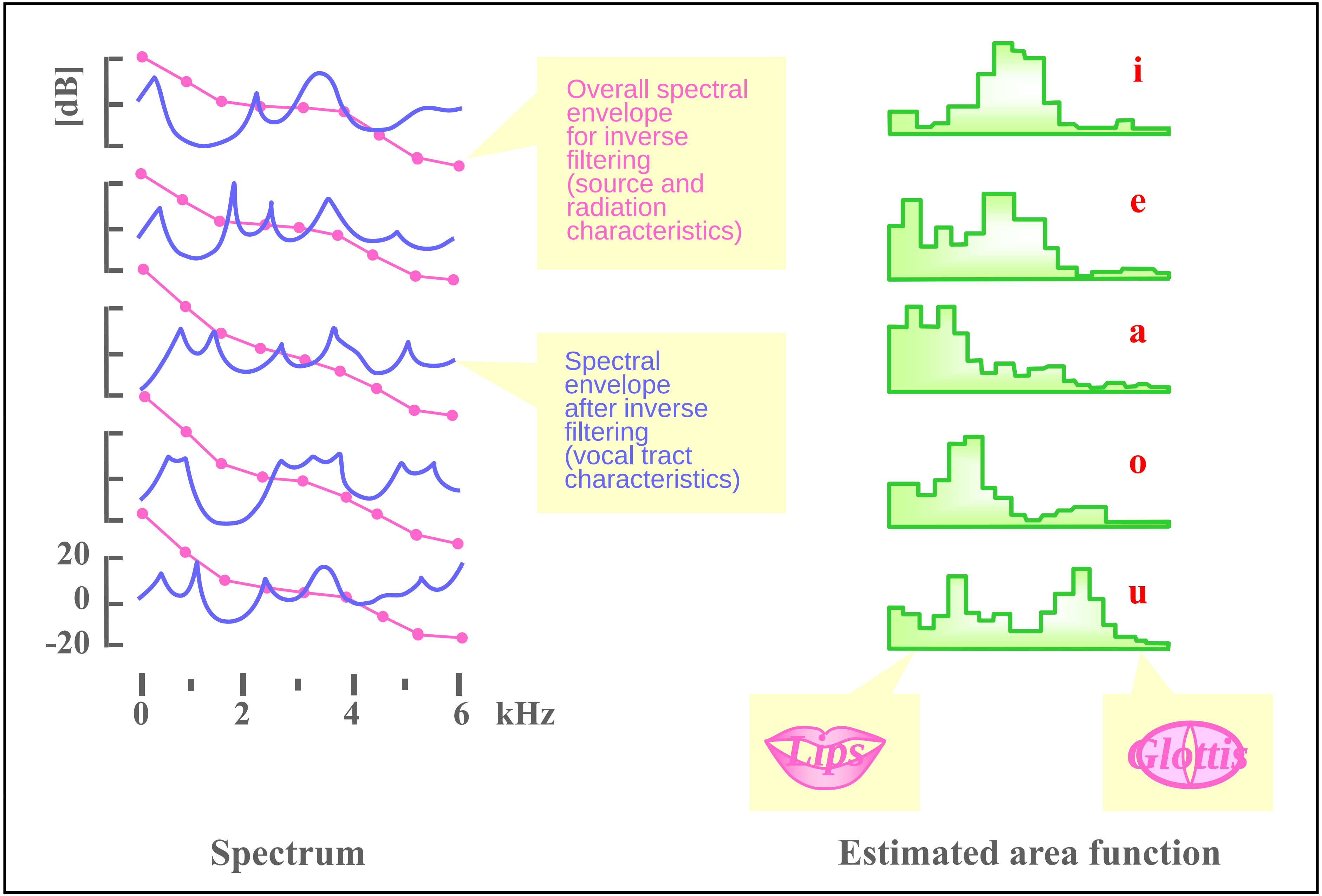
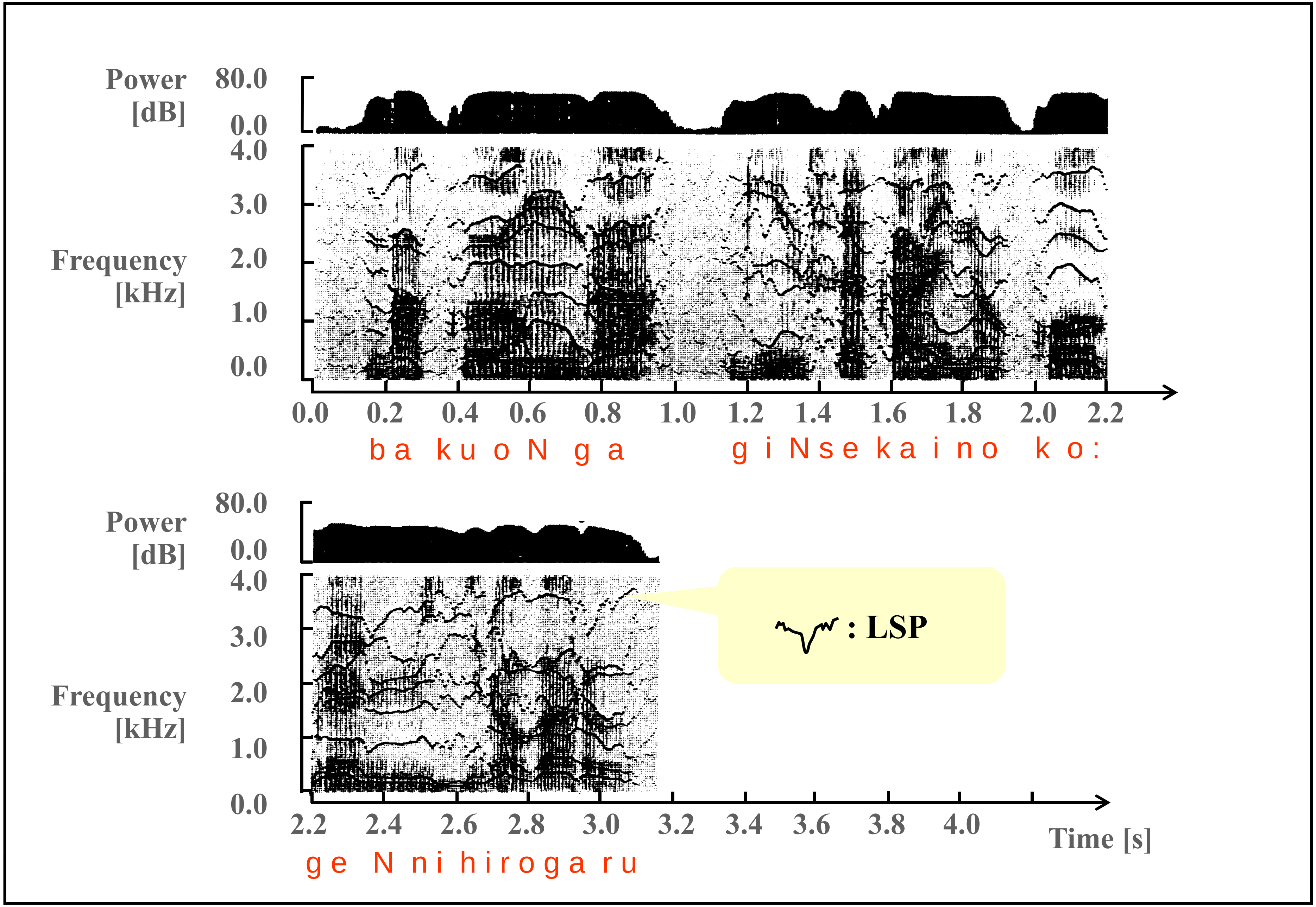
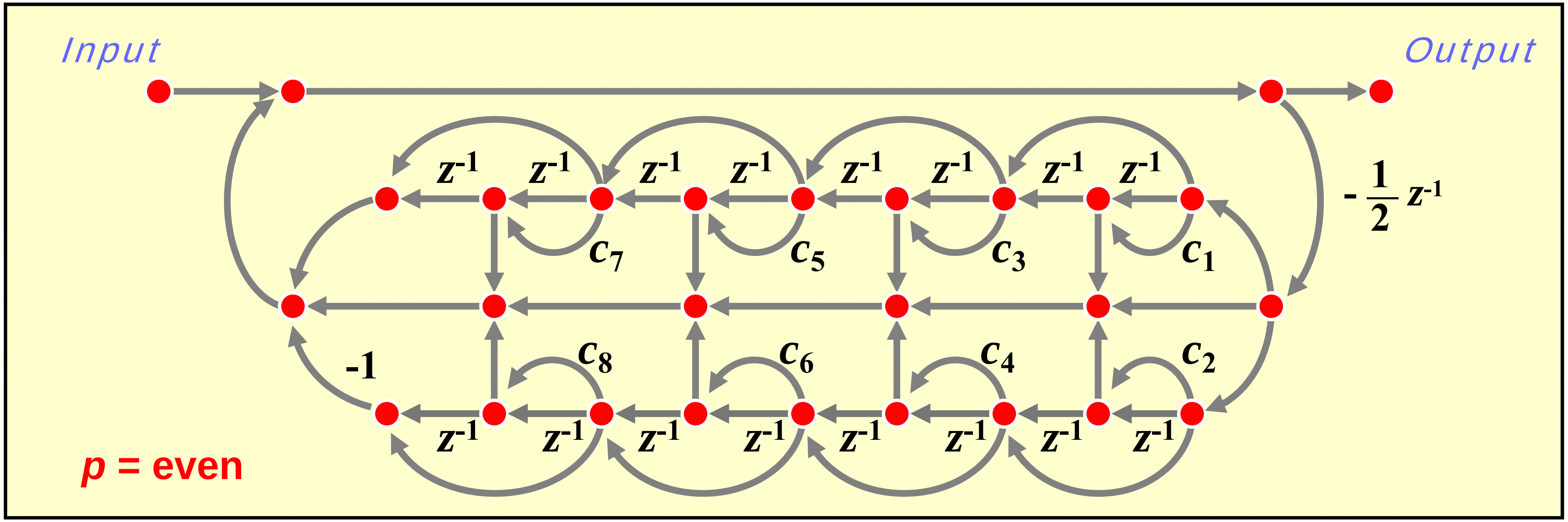
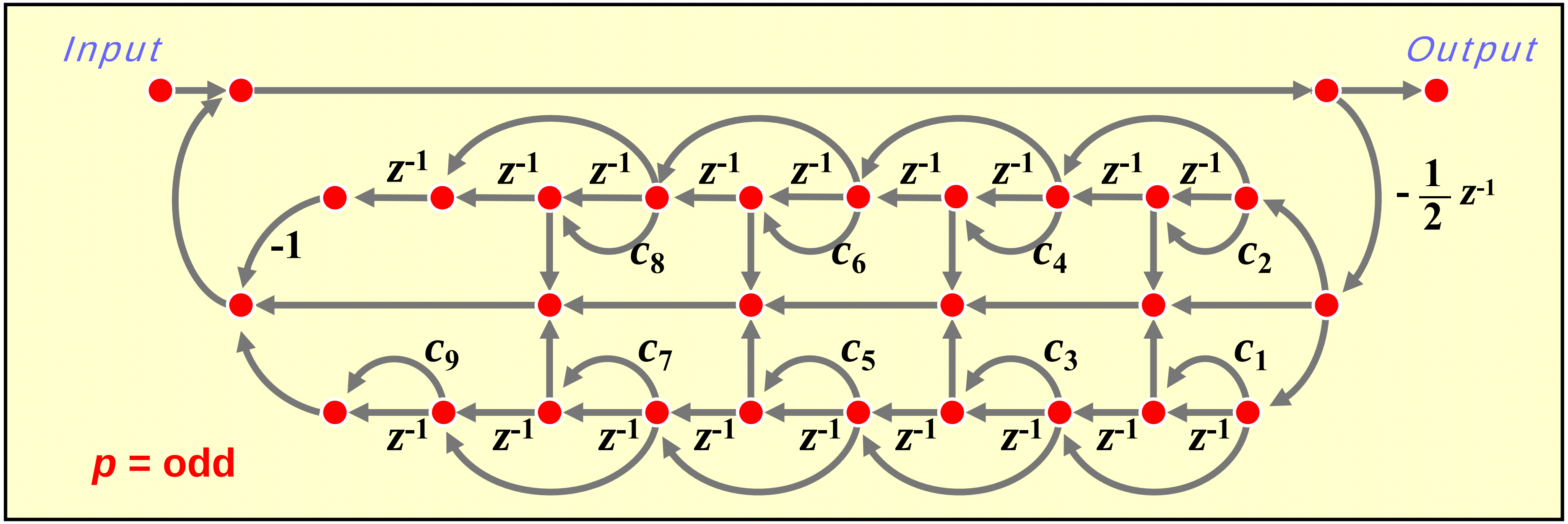


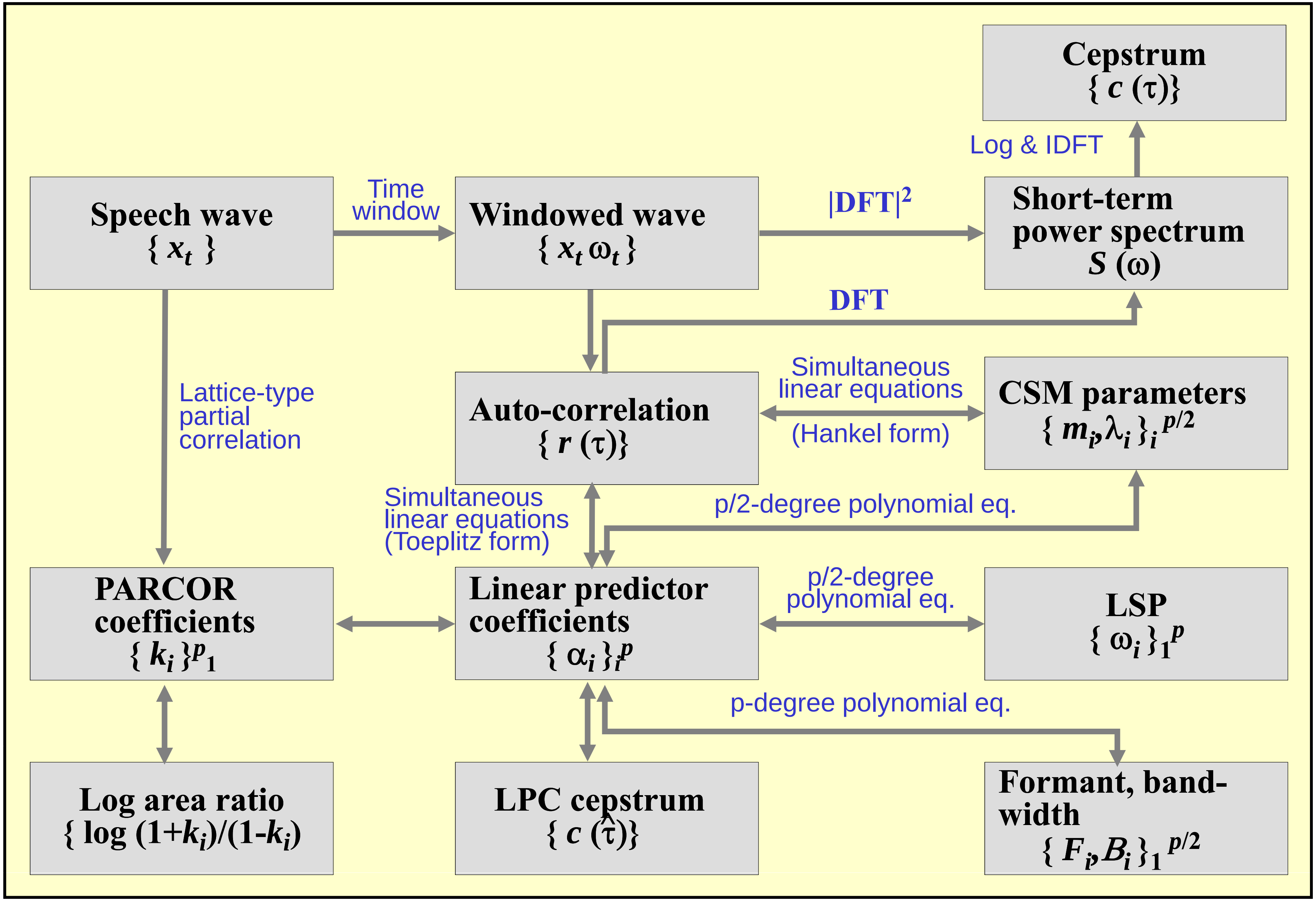


An example of LSP analysis





Signal flow graph of LSP synthesis filter



Mutual relationships between parameters based on all-pole spectrum modeling