G723.1 Standard

This coder is based on the same principles as the CELP coder, that is, it is based on the principles of linear predictive analysis-by-synthesis coding and attempts to minimize a perceptually weighted error signal. However it has some distinguishing features.

The G.723.1 coder operates on frames of 240 samples each (30 ms at an 8 kHz sampling rate). Each block is divided into 4 subframes of 60 samples each.

•For every subframe a 10th order linear prediction filter is computed using the original speech signal. The LPC are transformed to the LSP. The set of LSP for the last subframe is quantized using vector quantizer. Due to using vector quantization only 24 bits are required for LSP instead of 34 bits in the CELP coder.

LSP quantizer

$\mathbf{p}_n^T = [p_{1,n} p_{2,n} \cdots p_{10,n}]$ DC removed vector of LSPs

$\bar{\mathbf{p}}_n = b[\tilde{\mathbf{p}}_{n-1} - \mathbf{p}_{\mathbf{DC}}]$ Prediction from the previously decoded LSP vector, b=12/32

$\mathbf{e}_n = \mathbf{p}_n - \bar{\mathbf{p}}_n$ the residual LSP error.

3 codebooks (3,3,4 components of $\mathbf{e}_n$) of size 256 codewords each are used for quantization. We minimize the weighted error

$$E_{l,m} = (\boldsymbol{p}'_m - \tilde{\boldsymbol{p}}_{l,m})^T \boldsymbol{W}_m (\boldsymbol{p}'_m - \tilde{\boldsymbol{p}}_{l,m}), \quad \begin{matrix} 0 \le m \le 2 \\ 1 \le l \le 256 \end{matrix}$$

$$\tilde{\boldsymbol{p}}_{l,m} = \bar{\boldsymbol{p}}_m + \boldsymbol{p}_{DC_m} + \boldsymbol{e}_{l,m}, \quad \begin{matrix} 0 \le m \le 2 \\ 1 \le l \le 256 \end{matrix} \qquad \boldsymbol{p}' = \boldsymbol{p} + \boldsymbol{p}_{DC}$$

LSP dequantizer

The three sub-vectors $\{\boldsymbol{e}_{l_m,m,n}\}_{m=0..2}$ form $\tilde{\boldsymbol{e}}_{\boldsymbol{n}}$

$$\tilde{\mathbf{p}}_{\mathbf{n}} = \bar{\mathbf{p}}_{\mathbf{n}} + \tilde{\mathbf{e}}_{\mathbf{n}} + \mathbf{p}_{\mathbf{DC}}$$

A stability check is performed on the decoded LSP vector $\tilde{p}_{i+1,n} - \tilde{p}_{i,n} \geq \Delta_{\min} = 62.5\,\text{Hz}, 1 \leq i \leq 9$

Otherwise vector components are modified as

$$\tilde{p}_i = \tilde{p}_i - \Delta_{\min}/2 \qquad \tilde{p}_{i+1} = \tilde{p}_{i+1} + \Delta_{\min}/2$$

If after 10 iterations the condition of stability is not met, the previous LSP vector is used.

LSP interpolation

$$\tilde{\boldsymbol{p}}_{ni} = \begin{cases} 0.75\tilde{\boldsymbol{p}}_{\boldsymbol{n\text{-}1}} + 0.25\tilde{\boldsymbol{p}}_{\boldsymbol{n}}, i = 0 \\ 0.5\tilde{\boldsymbol{p}}_{\boldsymbol{n\text{-}1}} + 0.5\tilde{\boldsymbol{p}}_{\boldsymbol{n}}, i = 1 \\ 0.25\tilde{\boldsymbol{p}}_{\boldsymbol{n\text{-}1}} + 0.75\tilde{\boldsymbol{p}}_{\boldsymbol{n}}, i = 2 \\ \tilde{\boldsymbol{p}}_{\boldsymbol{n}}, i = 3 \end{cases}$$

G.723.1

Both the quantized and the unquantized LPC are used to construct the short-term perceptual weighting filter, which is used to filter the entire frame and to obtain the perceptually weighted speech signal.

•To speed up a search over the adaptive codebook the following approach is used. For every two sub-frames (120 samples), the open loop pitch period is computed :

$$\max_j \frac{\left(\sum_{i=0}^{119} s(n+i)\cdot s(n+i-j)\right)^2}{\sum_{i=0}^{119} s(n+i-j)^2}, 18 \le j \le 145$$

G723.1

Using the pitch predictor that is a linear FIR filter of 5th order given by its pulse response $(\alpha_{-2}, \alpha_{-1}, \alpha_0, \alpha_1, \alpha_2)$ the optimal prediction of the periodical part of the speech signal is computed by minimizing

$$\min_{j,\boldsymbol{\alpha}} = \sum_{n=0}^{59} \left(s(n) - \sum_{i=-2}^{i=2} \alpha_i s(n-p+i-j) \right)^2,$$

where $p = j_{opt}$ is the pitch estimate.

For subframes 0, 2 $j = \pm 1$ and $p + j$ is transmitted using 7 bits. For subframes 1,3 $p + j$ is coded differentially using 2 bits and may differ from the previous subframe pitch lag only by –1, 0, +1, or +2.

The coefficients $(\alpha_{-2}, \alpha_{-1}, \alpha_0, \alpha_1, \alpha_2)$ are vector quantized using two codebooks with 85 or 170 entries for high rate and 170 entries for low rate.

G723.1

The contribution of the pitch predictor $p(n), n = 0,...,59$ is subtracted from the target vector $s(n), n = 0,...,59$ to obtain the residual signal : $r(n) = s(n) - p(n), n = 0,...,59$

The non-periodic component of the excitation is approximated using the obtained residual signal. For the high bit rate, multi-pulse maximum likelihood quantization (MP-MLQ) excitation is used, and for the low bit rate, an algebraic codebook excitation (ACELP) is used.

The residual signal $r(n), n = 0,...,59$ is approximated as follows:

$$r'(n) = \sum_{j=0}^{n} h(j) \cdot v(n-j) \ , 0 \le n \le 59,$$

where $v(n)$ is the excitation of the filter with pulse response $h(n)$: $v(n) = G \sum_{m=0}^{M-1} \beta_m \delta(n - l_m) \ , 0 \le n \le 59$

G723.1

G is the gain factor, $\delta(n)$ is a Kronecker delta-function, β_m and l_m are the signs (± 1) and the positions of Kronecker functions, respectively, and M is the number of pulses.

$M = 6$ for even subframes and it is 5 for odd subframes.

The positions can be either all odd or all even. This is indicated by 1 bit.

The problem is to estimate the unknown parameters G, β_m, l_m,

$m = 0,..., M-1$ that minimize the squared error

$$\sum_{n=0}^{59}(err(n))^2 = \sum_{n=0}^{59}(r(n)-r'(n))^2 = \sum_{n=0}^{59}\left(r(n) - G\sum_{m=0}^{M-1}\beta_m h(n-l_m)\right)^2$$

G723.1

The following gain estimate is computed:

$$G_{\max} = \frac{\max_{j=0,1,\ldots,59}\left\{\left|\sum_{n=j}^{59} r(n)h(n-j)\right|\right\}}{\sum_{n=0}^{59}(h(n))^2}.$$

The estimated gain $G_{\max}$ is scalar quantized with cell size 3.2 dB (nonuniform quantizer with 24 cells). Around the approximating value, $\tilde{G}_{\max}$, additional gain values are selected within the range

$$[\tilde{G}_{\max} - 3.2, \tilde{G}_{\max} + 6.4]dB$$

For each of these gain values the signs and locations of the pulses are estimated and quantized. Finally, the combination of the quantized parameters that yields the minimum of $\sum_{n=0}^{59}(err(n))^2$ is selected. The best gain approximation and the optimal pulse locations and signs are transmitted.

Combinatorial coding

Combinatorial coding is used to transmit the pulse locations.

Assume that we consider binary sequences of length L and weight w it is easy to compute that there are

$N = \binom{L}{w}$ such sequences. The combinatorial coding spend

$\log_2 N$ bits to transmit the number of sequence.

For example, let $L = 30$ and $w = 6$ then $N = \binom{L}{w} = 593775$

and $\lceil \log_2 N \rceil = 20$ bits.

G723.1

The standard provides two bit rates 5.3 kb/s and 6.4 kb/s.

For rate 5.3 kb/s a 17-bit algebraic codebook is used to innovate excitation $v(n)$. The innovation vector contains at most 4 non-zero pulses. The amplitudes and positions are chosen from the admissible values given in the Table. The positions of all pulses can be simultaneously shifted by one (to occupy odd positions) which needs 1 extra bit.

Each pulse position is encoded with 3 bits and each pulse sign is encoded in 1 bit.

This gives a total of 16 bits for the pulses. An extra bit is used to encode the shift resulting in a 17-bit codebook.

ACELP pulse positions

Amplitude	Positions
±1	0, 8, 16, 24, 32, 40, 48, 56
±1	2, 10, 18, 26, 34, 42, 50, 58
±1	4, 12, 20, 28, 36, 44, 52, (60)
±1	6, 14, 22, 30, 38, 46, 54, (62)

ACELP

The codebook is searched by minimizing the MSE between the residual signal $r(n)$ obtained after subtracting the pitch contribution from the original speech, and the filtered weighted codeword $v(n)$ from the algebraic codebook

$$\sum_{n=0}^{59}\left(r(n)-G\sum_{j=0}^{n}h(j)v(n-j)\right)^2$$

where $h(n)$ is the pulse response of the filter, G is the codebook gain.

Bit allocation for the 6.4 kb/s coding algorithm

Parameters coded	Sub-frame 0	Sub-frame 1	Sub-frame 2	Sub-frame 3	Total
LPC indices					24
Adaptive codebook lags	7	2	7	2	18
All the gains combined	12	12	12	12	48
Pulse positions	20	18	20	18	76
Pulse signs	6	5	6	5	22
Grid index	1	1	1	1	4
TOTAL:					192

Bit allocation of the 5.27 kb/s coding algorithm

Parameters coded	Sub-rame 0	Sub-frame 1	Sub-frame 2	Sub-frame 3	Total
LPC indices					24
Adaptive codebook lags	7	2	7	2	18
All the gains combined	12	12	12	12	48
Pulse positions	12	12	12	12	48
Pulse signs	4	4	4	4	16
Grid index	1	1	1	1	4
TOTAL:					158

G.723.1

The major differences between two rates are in the pulse positions and amplitudes coding. Also at the lower rate only 170 codebook entries are used for the gain vector of the long term predictor.

Bit rate for highrate coder is: (8000/240)192=6.4 kb/s

Bit rate for lowrate coder is: (8000/240)158=5.27kb/s.

Other CELP-like coders

G.728 is a low-delay CELP. It uses a **backward adaptive** LP filter that is updated every 2.5 ms. The filter coefficients are not transmitted. At the decoder side the filter is constructed using synthesized speech signal.

The 10^{th} order gain predictor is used and there are 1024 excitation vectors which are split into 4 possible gains (quantized differences between gains and their predictions), 2 possible signs and 128 possible shapes. Excitation is transmitted for each frame containing 5 samples.

Bit rate is equal to $8000/5\times(2+1+7)=16$ kb/s.

IS-54 (VSELP), PDC(VSELP) coders. They represent vector sum excited linear prediction coders. Non-periodical excitation is linear combination of vectors from 2 highly structured codebooks.

Other CELP-like coders

GSM standard uses RPE-LTP (regular-pulse excitation- long-term prediction) coder. Operates at 13 kb/s. Uses frames of 160 samples and subframes of 40 samples. LP filter of order 8 is constructed. Its LSPs are encoded by 36 bits.

For each subframe periodical part is represented by gain and pitch. It requires (7+2)4=36 bits for representation.

Non-periodical part of the excitation is approximated as a sequence of multiple uniformly, i.e., (pitch-spaced) pulses. The residual is decimated into 3 subsequences of length 13 samples. Phase of the chosen group (with highest energy) is represented by 2 bits and maximum amplitude by 6 bits. It is approximated by a regular pulse sequence. Each pulse in this sequence has an amplitude quantized with 3 bits.

(13×3+2+6)×4=188 for excitation +36 for LSPs+36 for pitch part=260 bits ,i.e., 260/20ms=13kb/s

MELP standard

Typically the LPC vocoders use two voicing states: voiced and unvoiced (pulse or noise excitation)

VC are based on speech production model but do not use analysis-by-synthesis method.

Main features of MELP vocoder are

1. Converting LPC to LSP
2. Splitting the voice band into 5 subbands for noise/pulse mixed excitation
3. Shaping pulse excitation
4. Using aperiodic pulses for transition regions between voiced and unvoiced speech segments

Provides bitrate 2.4 kb/s

MELP

The encoder splits the input sequence into frames of size 180 samples (22.5 ms). The 10^{th} order LP filter is constructed based on the autocorrelation method and the LD procedure. The obtained LPCs are converted to LSPs.

Pitch is estimated as $R(p) = \dfrac{c_p(0, p)}{\sqrt{c_p(0,0)c_p(p, p)}}$

$$c_p(i, j) = \sum_{n=-p/2-80}^{p/2+80} s(n+i)s(n+j), \quad 40 \le p \le 160$$

MELP

The determination of voicing is performed for 5 subbands: 0-500 Hz, 500-1000 Hz, 1000-2000 Hz, 2000-3000 Hz, and 3000-4000 Hz. For this purpose the input speech is filtered by 6th order Butterworth bandpass filters. The encoder determines five voicing strengths vbp_i, $i = 1,...,5$ and refines pitch estimate p and the corresponding $R(p)$. The maximum autocorrelation value $R(p+\Delta)$ in the first subband determines the pitch estimate $p+\Delta$ and is used as vbp_1. If $R(p+\Delta) \le 0.5$ then aperiodic flag=1 otherwise it is set to 0 (decoder uses aperiodic pulse excitation or periodic with pulse period).

MELP

The residual signal is calculated by filtering the input speech by the found LP filter. The voicing flags in the other 4 subbands are calculated using the autocorrelation analysis and a peak search in the residual signal. If peakiness >1.34, $vbp_1 = 1$, if peakiness>1.6, $vbp_i = 1$, $i = 1,2,3$ 512-point FFT is performed on 200 samples of the residual signal and the first 10 harmonics with positions $512i / \hat{p}$, $i = 1,\ldots,10$ are selected, where $\hat{p}$ is the pitch estimate. A spectral peak-searching algorithm finds the maximum within $\lfloor 512 / \hat{p} \rfloor$ frequency samples around the initial position for each of 10 harmonics.

MELP

The amplitudes of the harmonics are vector quantized using a codebook of size 256.

In the decoder IDFT on one pitch length is performed.

Since only 10 amplitudes are transmitted and p amplitudes are required then the remaining amplitudes are set to 1. The single pulse excitation is

$$e_p(n) = p^{-1} \sum_{k=0}^{p-1} F(k) e^{j 2\pi k n / p}$$

$F(k)$ are the Fourier amplitudes. The noise excitation is generated by a random number generator.

MELP

The pulse and noise excitations are filtered and summed up to form the mixed excitation. The pulse excitation filter is given by the sum of all band-pass filter coefficients for the voiced subbands. It has the following transfer function

$$H_p(z) = \sum_{i=1}^{5} \sum_{k=0}^{6} v_i b_{i,k} z^{-k}$$

where $b_{i,k}$ are the filter coefficients and v_i are the quantized voicing strengths. The noise excitation is filtered by the filter with the transfer function

$$H_n(z) = \sum_{i=1}^{5} \sum_{k=0}^{6} (1 - v_i) b_{i,k} z^{-k}$$

, i.e. the sum of the bandpass filter coefficients for the unvoiced subbands.

MELP

Bit allocation of the MELP coder

LSPs	25	(4 codebooks:128+64+64+64)
Fourier amplitudes	8	
Two gains	8	
Pitch	7	
band voicing flags	4	
Aperiodic flag	1	
Sync bit	1	
Total	**54**	

Image coding standards

We use the term image for still pictures. Image coding includes compressing so-called bi-level or fax images, photographs (continuous-tone colour or monochrome images), and document images.

To compress image it is necessary to remove any redundancies observable in the signal. The obvious form of signal redundancy in most images is spatial redundancy.

Spatial redundancy takes a variety of different forms in an image, including correlations in the background image (e.g., a repeated pattern in a background wallpaper of a scene), correlations across an image (repeated occurrences of base shapes, colours, patterns, and so on).

Spatial redundancy

Repeated pattern on the background

Image coding

A variety of techniques used in modern image compression standards to compensate spatial redundancy is based on transform coding.

The second basic principle of image coding is to exploit the human's ability essentially to pay no attention to various types of image distortions. By understanding the masking properties of the human visual system it is possible to make distortions perceptually invisible.

Coding of bi-level fax images: JBIG Standards

Coding bi-level images

Each pixel of fax images can be white or black. These two pixel values can be represented by only one bit: 0 corresponds to black pixel and 1 corresponds to white pixel.

Group of successive pixels of the same colour we call run.

The binary representation of the given line is 100000010100.

G3 fax standard uses a 1-D run-length coding of pixels on each line followed by Huffman coding.

For our example we obtain (1,1) (6,0) (1,1) (1,0) (1,1) (2,0),

where the first element of the pair is a run length and the second is the pixel value. This sequence of pairs (run length,value) is coded by the 2-D Huffman code.

Coding bi-level images

The G4 fax standard provides (on average) an improvement over G3 by using 2-D run-length coding to take advantage of vertical spatial redundancy, as well as the horizontal spatial redundancy used in G3 fax coding.

G4 uses the pixels of the previous line as predicted values for the pixels of the current line. It is performed as follows. Starting with the second line the sum modulo 2 of previous and current lines is computed. The obtained sum is processed by run-length coder followed by a Huffman coder.

G3,G4 provide good compression for simple text-based faxes and they give small compression ratios for handwritten text or binary halftone images (gray-scale images converted into binary form).

Coding bi-level images

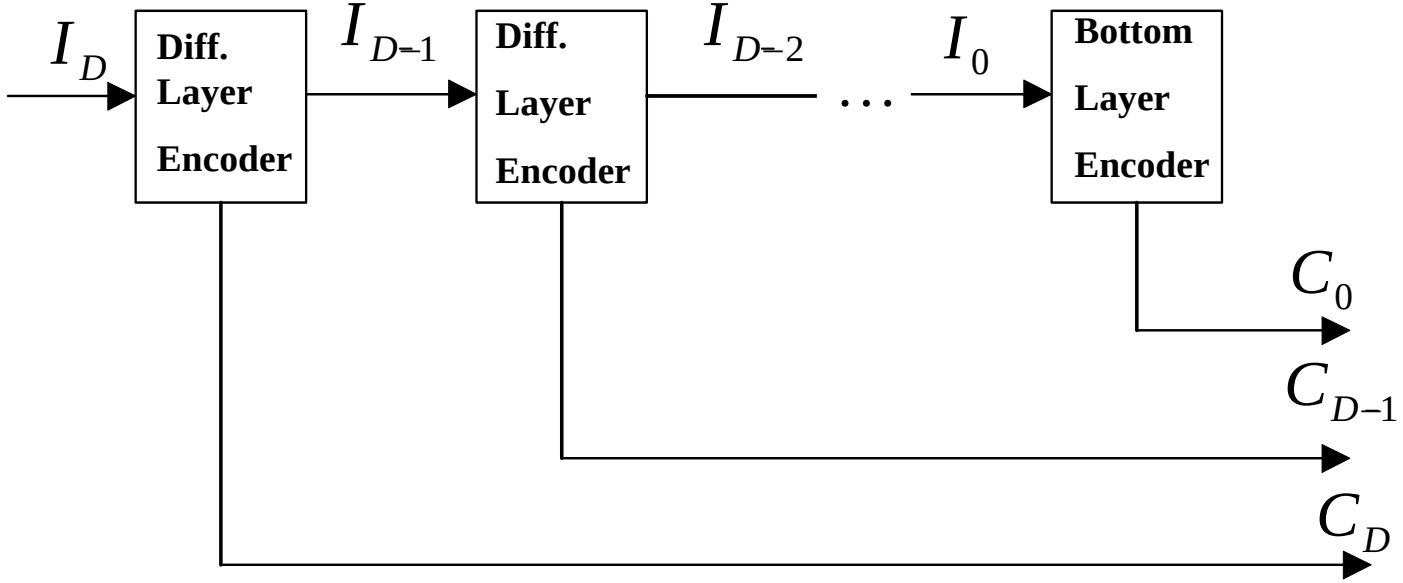
Binary halftone (gray-scale) images have statistical properties that are very different from binary text and therefore need a significantly different coding algorithm to provide high-quality encoding at significant compression ratios.

Halftone (gray-scale) images converted into dots (as in newspapers) contain rather short runs and need significantly larger region of support for prediction than that needed for text images. Halftone images represented by bit-planes contain rather short runs as well.

(Joint Bi-level Image experts Group) JBIG-1 standard provides compression ratios that are comparable to G4 for text sequences and it significantly improves compression for binary halftone images.

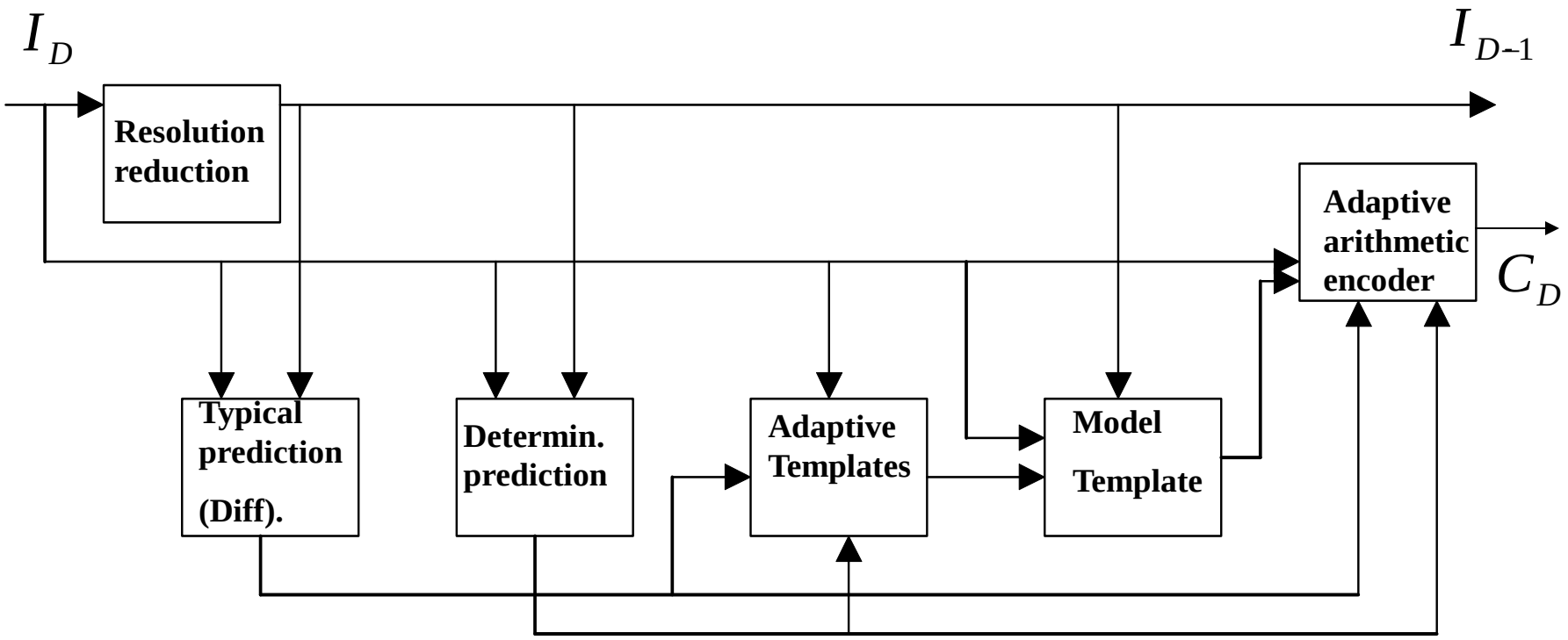
JBIG-1 standard

•JBIG-1 has a progressive mode that provides a low-resolution base-image and a sequence of delta files each of which corresponds to another level of resolution (each delta file doubles both the vertical and the horizontal resolution).



JBIG encoder

JBIG-1 standard



JBIG differential level

JBIG-1 standard

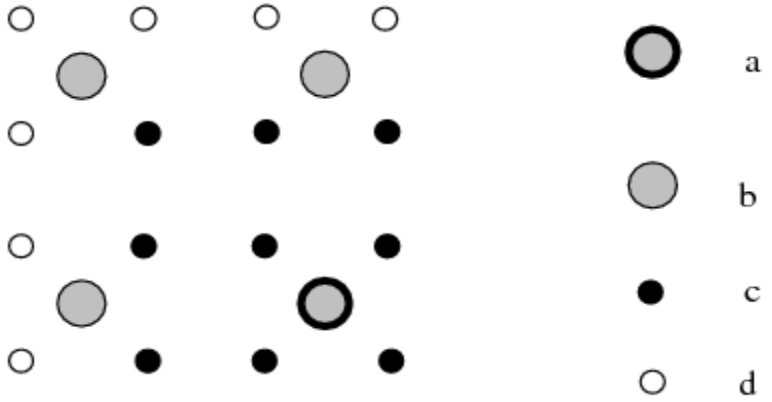
- The resolution reduction block decimates I_D to obtain I_{D-1}
- Remaining blocks implement the compression algorithm for I_D

The most important parts are an adaptive arithmetic coder and the model-templates block. They provide context arithmetic coding for pixels of I_D. For differential-layer coding context (an integer S) is determined by 6 pixels in the causal high-resolution image, 4 already available pixels in low-resolution image and also 4 possible spatial phase of the pixel to be encoded.

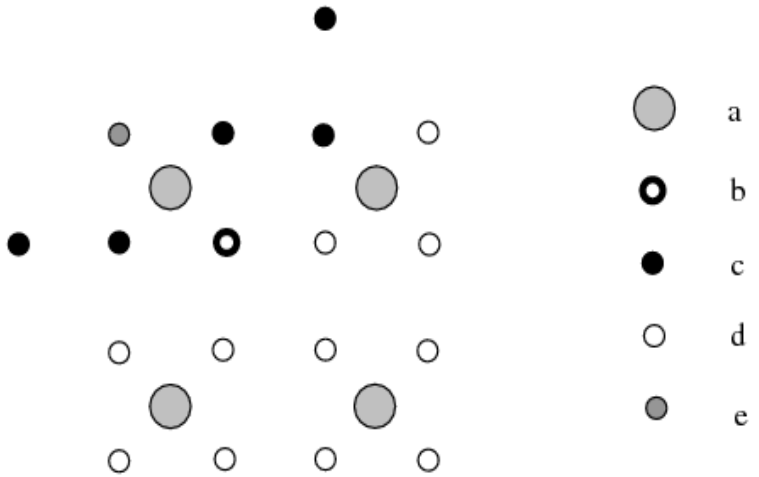
There are $4 \times 2^6 \times 2^4 = 4096$ possible contexts.

For bottom-layer encoder we have 1024 possible contexts which are formed by 9 adjacent already decoded pixels and 1 spatially separated from others.

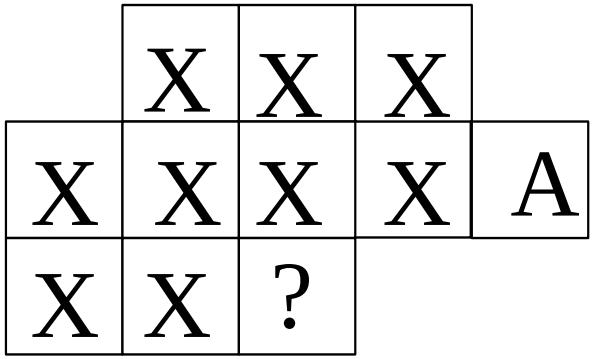
Resolution reduction



Context arithmetic coding



Context arithmetic coding



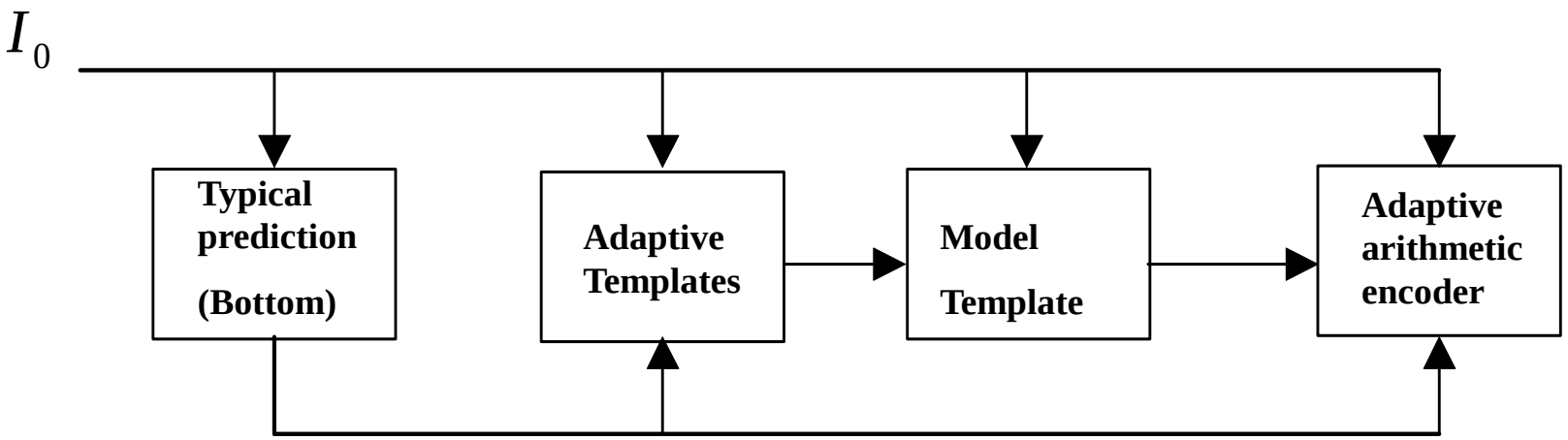
JBIG-1 standard

•**Adaptive templates block** finds periodicities in the image and chooses symbol “A” taking into account these periodicities.

•**Differential-layer typical prediction block** looks for regions of solid color and transmits it by a special flag. None of the processing normally is done in the deterministic prediction, adaptive templates, model templates or arithmetic coding blocks**.**

•**Bottom-layer typical predictor** also tries to exploit solid regions of image to save processing efforts. But it is line-skipping algorithm. A line is typical if it is identical to the line above. Flag “typical” is sent. The encoder and the decoder skip the coding of all pixels in typical lines and generate them by repetition.

JBIG-1 standard



JBIG bottom-layer encoder