## I-4. Outline of ADPCM Voice Analysis/Synthesis

The MSX-AUDIO has two different voice data processing functions:

adaptive differential pulse code modulation (ADPCM), which specifies voice analysis/synthesis and synthesizes realistic sound, and composite sinusoidal modeling, which requires complicated processing (using a computer) for data analysis, but less memory capacity for synthesis. ADPCM voice analysis/synthesis, which is the major feature of the MSX-AUDIO along with the FM sound generator, is outlined as follows:

ADPCM is a voice analysis/synthesis method in which the difference between actual data and expected data is encoded using a quantization width (adaptive quantization width) that changes according to waveform variation. This method prevents the synthesized voice from deterioration and reduces the number of required data bits. The encoding and decoding processes are described as follows:

## (a) Voice analysis

The MSX-AUDIO converts 8-bit PCM data into 4-bit ADPCM data.

- (1) Voice data of every sampling rate (1.8 KHz-16 KHz) is converted into 8-bit PCM data (X1,n).
- (2) The resulting PCM data (X1,n) is multiplied by 256 for conversion into 16-bit data. (Xn), and is then compared with the expected data  $(\widehat{X}n)$  to obtain a difference (dn).
- (3) When the difference is a positive value, ADPCM data MSB(L4) is specified as "0"; when negative, as "1". At the same time, the absolute value (|dn|) of the difference is caluculated.
- (4) Then, the remaining three bits are determined by the relationship between the absolute value (|dn|) and quantization width ( $\Delta$  n) as shown in Table I-1.

Table I-1 ADPCM codes

## 1. Condition

L.		L <sub>3</sub>		L,	*	(1)
dn ≧ 0	dn < 0	L3	L <sub>2</sub>	Lı	^	
0	1	0	0	0		$dn \bot \leq \triangle n                               $
		0	0	1	∆n.⁄4≦	$d\mathbf{n}^{+} \leq \triangle \mathbf{n} \mathbb{Z}_{2}$
		0	1	0	∆n⁄2≦	dn  <∧n∗¾
		0	1	1	∆n≠%≤	$dn^{\perp} \leq \triangle n$
		1	0	0	∆n≨	$dn \bot < \triangle n * \%$
		1	0	1	∆n*%≦	$dn \vdash < \triangle n * \%$
		1	1	0	∆n <b>*</b> ½ ≦	dn! < △n <b>*</b> %
		1	1	1	∆n∗¾≨	dn i

Conversion of voice data into ADPCM data is thus completed.

(5) After ADPCM data has been obtained, the new expected data  $(\hat{X}n+1)$  and quantization width  $(\Delta n+1)$  are created.