CONTENTS

Q1: What kind of method is ADPCM?

A1 First the basic DM and ADM methods are mentioned, followed by the ADPCM method.

• DM and ADM methods

In the DM (Delta Modulation) method a certain quantity (Δ) predetermined for each cycle of sampling is added or subtracted to express a voice waveform. In other words, the addition of a Δ is encoded as 1, while the subtraction of a Δ is encoded as 0. Thus, rapid changes in voice waveform with respect to a step width Δ cannot be covered by this method. Figure 10.1 (a) illustrates this case.

For the ADM (Adaptive Delta Modulation) method, the quantized width (Δ) is adjusted according to the rapid changes to improve the response of the voice waveform. Encoding one bit is based on this means. If a value of 1 or 0 continues for a certain period, the quantized width (Δ) is enlarged for quicker response. Figure 10.1 (b) illustrates such quick response.

• ADPCM method

In the ADPCM (Adaptive Differential Pulse Code Modulation) method is such that the basic width of quantization (Δ) is adjusted adaptive to the rapid changes for each cycle of sampling to encode each signal as three to four bits of data. This provides higher response of voice waveform.

For example, in the case of four-bit ADPCM, the upper one bit stands for polarity (increase or decrease) data, while the lower three bits determine the multiplier factor to the basic width of quantization (Δ). (The Δ value depends on the correlation with the past data.) This means that about 400 pieces of data [(about 50 Δ pieces) x 8 (for 3 bits)] can be changed at a time. Therefore, approximately 400 pieces of data are available. Figure 10.2 illustrates this case.

Furthermore, in the three-bit ADPCM, the upper one bit stands for polarity (increase or decrease) data, while the lower two bits determine the multiplier factor to the basic width of quantization (Δ). This means that about 200 pieces of data [(about 50 Δ pieces) x 4 (for 2 bits)] can be changed at a time. Therefore, 200 pieces of data are available.

The ADPCM method allows the achievement of high-quality sound in a relatively simple configuration, and the easy creation of voice data.

Note, however, that there is no compatibility between ADPCM data created by different manufacturers.

(Note) When playback starts from part of a phrase, normal waveforms cannot be reproduced because the ADPCM algorithm is employed.
See Q14 and A14.

Figure 10.1
DM and ADM
Methods

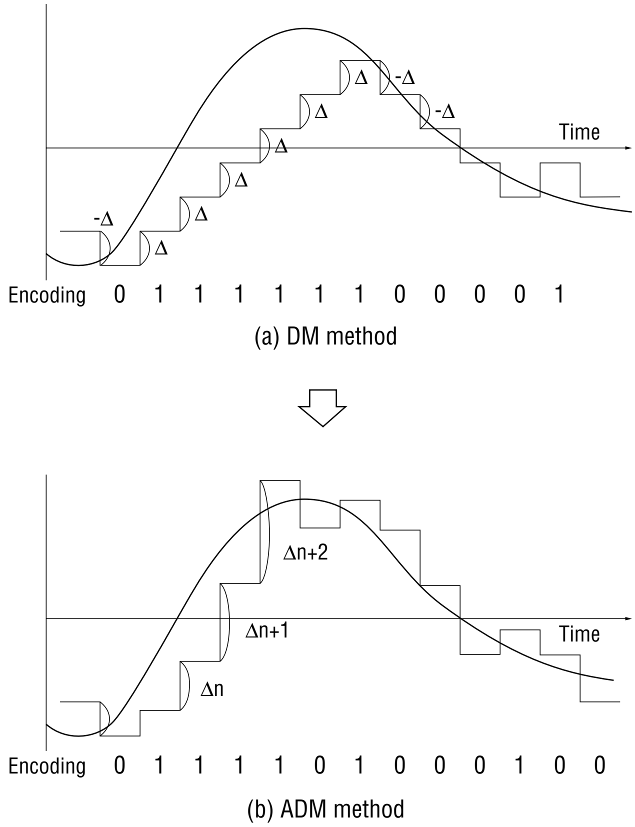
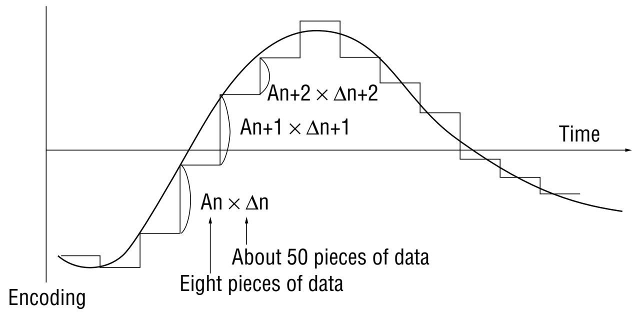


Figure 10.2
4-bit ADPCM



Polarity bit	0	1	1	1	1	1	0	0	0	0	0	0
Multiplier factor to Δ (An)	0	0	1	1	0	0	0	0	0	1	0	0
	1	1	0	1	0	0	1	1	1	0	1	0
	0	1	0	0	1	1	0	0	1	0	0	0

Q2: What is the specific value of the quantized width by ADPCM?

A2 The quantized width is an important factor in determining sound quality implemented by the ADPCM method, and is the know-how owned by individual manufacturers. Hence, detailed data of our ADPCM method (OKIADPCM) cannot be disclosed. The OKIADPCM method is not compatible with the ADPCM method conforming to ITU-T (former CCITT).

Q3: How should we perceive the relationship between bit rates and the synthesis length and between bit rates and sound quality?

A3 Bit rate and length of synthesis

A bit rate indicates the degree of information compression, and how many bits of data are required for synthesis in one second. Thus, the bit rate depends on the sampling frequency and the amount of data per sample, and is determined by the following formula:

Bit rate (kbps) = sampling frequency (kHz) x amount of data per sample (bits)

Example 1

Sampling frequency: 4 kHz, data: 4-bit ADPCM

Bit rate (kbps) = 4 (kHz) x 4 (bits) = 16 (kbps)

Example 2

Sampling frequency: 8 kHz, data: 4-bit ADPCM

Bit rate (kbps) = 8 (kHz) x 4 (bits) = 32 (kbps)

The length of synthesis depends on the memory capacity and the bit rate, as shown in the following formula:

$$\text{Synthesis length (seconds)} = \frac{1.024 \times \text{(memory capacity) (Kbits)}}{\text{bit rate (kbps)}}$$

$$= \frac{1.024 \times \text{(memory capacity) (Kbits)}}{\text{sampling frequency (kHz)} \times \text{data amount per sample (bits)}} \text{ (seconds)}$$

Example

Sampling frequency: 4 kHz, 4-bit ADPCM, Memory capacity: 256 Kbits

$$\text{Synthesis length (seconds)} = \frac{1.024 \times 256 \text{ Kbits}}{4 \text{ (kHz)} \times 4 \text{ (bits)}} \fallingdotseq 16.4 \text{ (seconds)}$$

Bit rate and sound quality

A lower bit rate results in a longer synthesis length. But the response to a voice waveform becomes lowered, with deteriorated sound quality.

Recording/Playback Time Quick Reference Table

The following is a quick reference table for recording/playback times calculated on the basis of each voice synthesis method.

Voice synthesis method	Bit length (bit)	Sampling frequency (kHz)	Bit rate (kbps)	Memory capacity							
				256Kbit	512Kbit	1Mbit	1.5Mbit	2.0Mbit	3.0Mbit	4.0Mbit	8.0Mbit
				Maximum playback time (second)							
ADPCM	4	4.0	16.0	16	33	66	98	131	197	262	524
		5.3	21.2	12	25	49	74	99	148	198	396
		6.4	25.6	10	20	41	61	82	123	164	328
		8.0	32.0	8	16	33	49	66	98	131	262
		10.6	42.4	6	12	25	37	49	74	99	198
		12.8	51.2	5	10	20	31	41	61	82	164
		16.0	64.0	4	8	16	25	33	49	66	131
		32.0	128.0	2	4	8	12	16	25	33	66
	3	4.0	12.0	22	44	87	131	175	262	350	699
		5.3	15.9	16	33	66	99	132	198	264	528
		6.4	19.2	14	27	55	82	109	164	218	437
		8.0	24.0	11	22	44	66	87	131	175	350
		10.6	31.8	8	16	33	49	66	99	132	264
		12.8	38.4	7	14	27	41	55	82	109	218
		16.0	48.0	5	11	22	33	44	66	87	175
		32.0	96.0	3	5	11	16	22	33	44	87
PCM	8	4.0	32.0	8	16	33	49	66	98	131	262
		5.3	42.4	6	12	25	37	49	74	99	198
		6.4	51.2	5	10	20	31	41	61	82	164
		8.0	64.0	4	8	16	25	33	49	66	131
		10.6	84.8	3	6	12	19	25	37	49	99
		12.8	102.4	3	5	10	15	20	31	41	82
		16.0	128.0	2	4	8	12	16	25	33	66
		32.0	256.0	1	2	4	6	8	12	16	33
(Reference) SBC	—	6.0	7.5	35	70	140	210	280	419	559	1118
			9.5	28	55	110	166	221	331	442	883
			12.0	22	44	87	131	175	262	350	699
		8.0	10.0	26	52	105	157	210	315	419	839
			12.6	21	42	83	125	166	250	333	666
			16.0	16	33	66	98	131	197	262	524

Q4.1: How can sinusoidal waves be produced by ADPCM?

A4.1 Tables 10.1.1 to 10.1.4 show ADPCM data for 4-bit (0 to F) sinusoidal waves, and Table 10.1.5 shows ADPCM data for 3-bit (0 to 7) sinusoidal waves. There are four kinds of 4-bit data corresponding to four, six, ten and twelve samples depending on the repeated data count per period of a sinusoidal wave. Figure 10.3 shows a model of an output waveform with a supply voltage (V_{DD}) of five volts.

See Q4.3, Table 10.2 for determining the frequency of the sinusoidal wave and the frequencies.

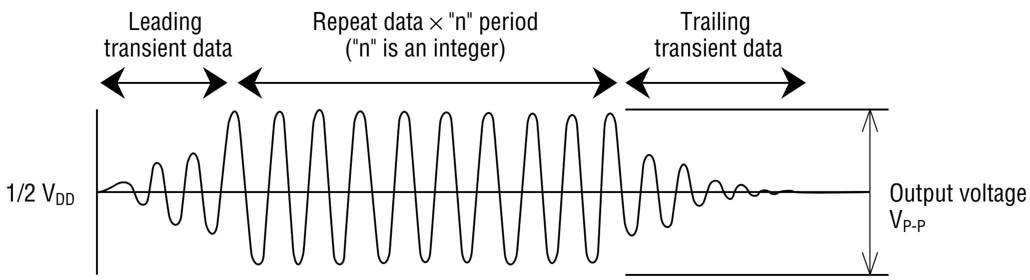


Figure 10.3 Sinusoidal Waveform Output by ADPCM

Table 10.1.1 Four-sample Sinusoidal Wave Output Voltage and ADPCM (4-bit) Data

Simusoidal wave frequency 2 kHz (f_{sam} = 8 kHz)

Output voltage (V_{DD} = 5 V)	Leading transient data	Repeat data	Trailing transient data
40% (2.0 V_{P-P})	1, 2, C, D, 4, 4, C, C, 4, 5, C, C, 4, 4, C, C, 4, 4, D, 9, 4, 0, C, 9, 4, 1, C, 8, 4, 0, C, 8, 4, 0, C, 8	4, 0, C, 8	2, 2, A, A, 2, 2, A, A, 2, 2, A, A, 2, 2, A, A, 2, 1, A, 9, 2, 1, A, 9, 2, 0, 8, 8, 0, 0, 8, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8
52% (2.6 V_{P-P})	1, 2, C, D, 4, 4, C, C, 4, 5, C, C, 4, 4, C, C, 4, 4, D, 9, 5, 0, D, 8, 5, 0, D, 8	5, 0, D, 8	4, 0, B, 8, 3, 0, B, 8, 3, 0, A, 8, 3, 0, B, 8, 2, 0, A, 8, 2, 0, A, 8, 2, 0, A, 8, 2, 0, A, 8, 2, 0, A, 8, 2, 0, A, 8, 1, 0, 8, 8, 8, 0, 8, 0, 8, 0
60% (3.0 V_{P-P})	1, 2, C, D, 4, 4, C, C, 4, 5, C, C, 4, 4, C, C, 4, 4, D, 9, 6, 0, E, 8	6, 0, E, 8	4, 0, B, 8, 3, 0, B, 8, 3, 0, A, 8, 3, 0, B, 8, 2, 0, A, 8, 2, 0, A, 8, 2, 0, A, 8, 2, 0, A, 8, 2, 0, 8, 0, 8, 8, 0, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8
70% (3.5 V_{P-P})	1, 2, C, D, 4, 4, C, C, 4, 5, C, C, 4, 4, C, C, 4, 4, D, 8, 7, 0, F, 8	7, 0, F, 8	5, 0, B, 8, 3, 0, A, 8, 2, 0, A, 8, 2, 0, A, 8, 2, 0, A, 8, 2, 0, A, 8, 2, 0, A, 8, 1, 0, A, 9, 1, 0, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8

Table 10.1.2 Six-sample Sinusoidal Wave Output Voltage and ADPCM (4-bit) Data

Simusoidal wave frequency 1.33 kHz (f_{sam} = 8 kHz)

Output voltage (V_{DD} = 5 V)	Leading transient data	Repeat data	Trailing transient data
17% (0.85 V_{P-P})	0, 0, 0, B, B, B, 7, F, 7, F, 4, C, 0, 0	4, 0, 9, C, 8, 1	3, 0, 8, C, 8, 1, 1, 1, 8, A, 8, 0, 1, 0, 8, 9, 8, 1, 1, 0, 9, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8
20% (1.0 V_{P-P})	0, 0, 0, 0, A, B, 7, F, 7, F, 4, D, 2, 9	4, 0, 9, C, 8, 1	3, 0, 8, C, 8, 1, 1, 1, 8, A, 8, 0, 1, 0, 8, 9, 8, 2, 9, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0
30% (1.5 V_{P-P})	0, 0, 0, 3, 3, 3, 7, F, 7, F, 4, C, 4, C	4, 0, 9, C, 8, 1	3, 0, 8, B, 8, 1, 2, 0, 9, B, 9, 3, 1, 1, 9, A, 8, 1, 1, 8, 0, 0, 9, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8
40% (2.0 V_{P-P})	0, 0, 0, 0, 4, A, 7, E, 7, E, 7, F, 8, 1	4, 0, 9, C, 8, 1	2, 0, 8, A, 8, 1, 1, 0, 8, 9, 8, 0, 1, 9, 1, 8, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0
52% (2.6 V_{P-P})	0, 0, 0, 0, 7, F, 7, F, 7, F, 4, C, 0, 0	4, 0, 9, C, 8, 1	3, 0, 8, B, 8, 1, 2, 0, 9, B, 8, 1, 2, 0, 9, A, 8, 1, 1, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0
70% (3.5 V_{P-P})	0, 0, 0, 0, 7, F, 7, F, 7, F, 4, C, 0, 0	5, 0, A, D, 8, 2	3, 0, 8, B, 8, 1, 2, 0, 9, B, 8, 1, 2, 0, 9, A, 8, 1, 1, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0
85% (4.25 V_{P-P})	0, 0, 0, 0, 7, F, 7, F, 7, F, 4, C, 0, 0	6, 0, B, E, 8, 3	3, 0, 8, B, 8, 1, 2, 0, 9, B, 8, 1, 2, 0, 9, A, 8, 1, 1, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0, 8, 0

Table 10.1.3 Ten-sample Sinusoidal Wave Output Voltage and ADPCM (4-bit) Data

Simusoidal wave frequency 800 Hz (f_{sam} = 8 kHz)

Output voltage (V_{DD} = 5 V)	Leading transient data	Repeat data	Trailing transient data
20% (1.0 V_{P-P})	7, 7, 7, 3, E, B, B, A, 0, 5	3, 3, 2, 8, D, B, B, A, 0, 5	3, 1, 0, 8, B, A, A, 8, 0, 3, 3, 1, 1, 8, A, 9, 9, 8, 0, 1, 2, 0, 0, 8, 8, 0, 8, 0, 8, 0
40% (2.0 V_{P-P})	0, 5, 7, 7, 4, E, F, B, 0, 4	3, 3, 2, 8, D, B, B, A, 0, 5	3, 1, 1, 8, B, B, A, A, 0, 4, 3, 2, 1, 8, B, B, A, 9, 1, 3, 3, 2, 1, 8, B, A, A, A, 1, 2, 1, 2, 8, 8, 0, 8, 8, 0, 8, 0, 8
60% (3.0 V_{P-P})	0, 6, 7, 7, 4, F, F, B, 0, 4	3, 3, 2, 8, D, B, B, A, 0, 5	3, 2, 0, 9, B, B, 9, 8, 2, 3, 3, 1, 0, 8, B, B, A, 8, 1, 3, 2, 1, 0, 8, A, A, 8, 8, 0, 2, 0, 0, 8, 8, 8, 0, 8, 0, 8, 0, 8, 0
80% (4.0 V_{P-P})	0, 7, 7, 7, 7, D, E, C, 0, 3	5, 2, 1, 9, A, D, A, 9, 1, 2	3, 2, 1, 9, 9, B, 9, 9, 1, 2, 3, 0, 0, 8, 9, B, 9, 9, 1, 2, 2, 0, 0, 8, 8, A, 9, 8, 0, 2, 1, 0, 0, 8, 8, 9, 8, 0, 0, 0, 8, 0, 8, 0

Table 10.1.4 Twelve-sample Sinusoidal Wave Output Voltage and ADPCM (4-bit) Data

Simusoidal wave frequency 667 Hz (f_{sam} = 8 kHz)

Output voltage (V_{DD} = 5 V)	Leading transient data	Repeat data	Trailing transient data
20% (1.0 V_{P-P})	7, 7, 7, 3, 9, D, C, B, A, 0, 1, 4	4, 3, 2, 8, 9, C, C, B, A, 0, 1, 4	3, 2, 1, 8, 9, A, A, A, 9, 0, 1, 2, 3, 1, 8, 9, A, 8, 9, 0, 1, 1, 0, 0, 0, 8, 0, 8
40% (2.0 V_{P-P})	7, 7, 7, 7, 1, C, C, B, B, 1, 1, 4	4, 3, 2, 8, 9, C, C, B, A, 0, 1, 4	3, 2, 0, 9, 8, A, A, A, 8, 0, 1, 2, 2, 1, 1, 0, 9, 8, A, 9, 9, 1, 1, 0, 0, 8, 0, 8, 0, 8, 8, 0, 8, 0
60% (3.0 V_{P-P})	7, 7, 7, 7, 4, C, D, B, A, 0, 1, 4	4, 3, 2, 8, 9, C, C, B, A, 0, 1, 4	4, 1, 0, A, 8, 8, A, A, 9, 2, 2, 1, 3, 2, 2, A, A, 9, B, B, B, 3, 2, 2, 3, 2, 1, 0, B, B, B, 9, 8, 8, 1, 1, 8, 0, 8, 0, 8, 0
80% (4.0 V_{P-P})	7, 7, 7, 7, 7, B, C, C, A, 0, 1, 4, 3, 3, 3, 8, 9, D, C, A, B, 0, 1, 4	4, 3, 2, 8, 9, C, C, B, A, 0, 1, 4	4, 2, 8, 9, A, A, 9, 9, 0, 1, 3, 3, 2, 1, 8, 9, A, B, A, 9, 0, 1, 3, 2, 2, 1, 9, 9, A, A, A, A, 0, 1, 3, 2, 0, 9, 8, 8, 0, 8, 0, 8, 0, 8

Table 10.1.5 Six-sample Sinusoidal Wave Output Voltage and ADPCM (3-bit) Data

Simusoidal wave frequency 1.33 kHz (f_{sam} = 8 kHz)

Output voltage (V_{DD} = 5 V)	Leading transient data	Repeat data	Trailing transient data
20% (1.0 V_{P-P})	3, 3, 3, 7, 7, 0, 3, 1, 5, 7	5, 1, 2, 1, 5, 6	5, 1, 5, 2, 1, 0, 4, 5, 4, 0, 0, 0, 4, 4, 0, 4, 0, 0, 4, 0, 4, 4, 0, 0, 4, 0, 4, 0, 4, 0, 4, 4, 0, 4, 0, 4, 0
40% (2.0 V_{P-P})	3, 3, 3, 7, 7, 5, 3, 3, 4, 7	5, 1, 2, 1, 5, 6	5, 1, 1, 1, 1, 5, 4, 4, 0, 0, 0, 4, 4, 0, 4, 0, 0, 4, 4, 0, 4, 0, 0, 4, 4, 0, 4, 0, 0, 4, 4, 0, 4, 0, 0, 4, 4, 0, 4, 0, 4
60% (3.0 V_{P-P})	1, 1, 1, 1, 1, 1, 3, 7, 3, 7, 3, 7, 3, 7, 3, 7	2, 0, 4, 6, 4, 0	1, 1, 4, 5, 5, 0, 1, 0, 4, 5, 5, 1, 1, 1, 4, 5, 4, 0, 1, 1, 5, 5, 5, 0, 1, 1, 4, 5, 5, 1, 1, 0, 5, 5, 4, 1, 0, 1, 5, 5, 4, 1, 0, 4, 0, 4, 0, 4

Q4.2: Can sinusoidal waves by ADPCM be used to produce an output voltage of 2.5 V_{p-p}?

A4.2 Data not covered in Tables 10.1.1 to 10.1.5 cannot be generated.

Q4.3: What are the frequencies of sinusoidal waves?

A4.3 f [kHz] stands for the frequency of a sinusoidal wave resulting from ADPCM data covered in Tables 10.1.1 to 10.1.5, while f_{SAM} [kHz] stands for the sampling frequency. If n samples of ADPCM data are repeated, the frequency of the resulting sinusoidal wave is given by the formula below.

$f = f_{SAM}/n$ (Common to four-bit and three-bit data.)

For actual values, see Table 10.2.

Table 10.2 Frequency of Synthesized Sinusoidal Wave

Sampling frequency (Hz) \ Repeated data samples	Frequency (Hz) of synthesized sinusoidal wave			
	4 samples	6 samples	10 samples	12 samples
32. 0 k	8. 0 k	5. 33 k	3. 2 k	2. 67 k
25. 6 k	6. 4 k	4. 27 k	2. 56 k	2. 13 k
24. 0 k	6. 0 k	4. 0 k	2. 4 k	2. 0 k
21. 2 k	5. 3 k	3. 53 k	2. 12 k	1. 77 k
16. 0 k	4. 0 k	2. 67 k	1. 6 k	1. 33 k
14. 0 k	3. 5 k	2. 33 k	1. 4 k	1. 17 k
12. 8 k	3. 2 k	2. 13 k	1. 28 k	1. 07 k
12. 0 k	3. 0 k	2. 0 k	1. 2 k	1. 0 k
10. 6 k	2. 65 k	1. 77 k	1. 06 k	883
10. 0 k	2. 5 k	1. 67 k	1. 0 k	833
8. 0 k	2. 0 k	1. 33 k	800	667
6. 4 k	1. 6 k	1. 07 k	640	533
6. 0 k	1. 5 k	1. 0 k	600	500
5. 3 k	1. 33 k	833	530	442
4. 0 k	1. 0 k	667	400	333

Q5: Why does voice synthesis require a low-pass filter?

A5 A low-pass filter (LPF) only passes the frequency components below a certain frequency in an input signal.

Synthesized voice output is issued by a DA converter. Therefore, a stepwise waveform is output, as shown below.

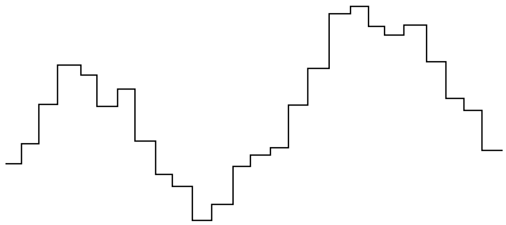


Figure 10.4 Voice Output Waveform from DA Converter

The waveform contains high-frequency noise components (metal-like sound). Moreover, the sampling theorem indicates that only the frequency components below half the sampling frequency are valid as the output. If high-frequency noise components are removed by an LPF, the following output waveform is obtained. (Smooth sound)

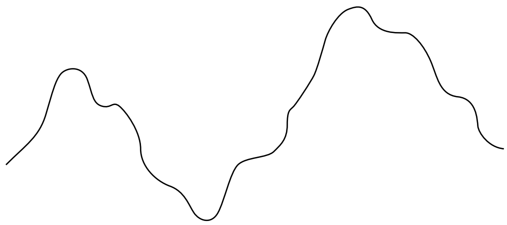


Figure 10.5 LPF Output Waveform

Q6: How should a cut-off frequency for a low-pass filter be determined?

A6 An ideal low-pass filter completely cuts off the frequency components above a certain frequency. In reality however, the output of frequency components above a certain frequency is attenuated in a slope.

Conventionally, such a slope is expressed as "dB/oct". For example, a slope of –12 dB/oct indicates that, when the frequency is doubled (an octave), the output becomes –12 dB (1/4).

The frequency at which attenuation is started is referred as a cut-off frequency (*1). The optimum cut-off frequency depends on the sampling frequency, attenuation characteristics, and frequency components of source voice. The conventional measure of design is the degree of attenuation at half the sampling frequency (f_{SAM}).

Table 10.3 lists reference values. As optimum value varies significantly with the frequency components of source voice, it is recommended that the cut-off frequency be determined in an auditory way.

Table 10.3 Relationship between Filter's Attenuation Characteristics, Cut-off Frequency and Sampling Frequency

Filter's attenuation characteristics (dB/oct)	Cut-off frequency	Gain at 0.5 f_{SAM} (dB)	Figure number
–12	0.3 f_{SAM}	–8.7	①
–18	0.33 f_{SAM}	–11	—
–24	0.35 f_{SAM}	–12.5	②
–48	0.38 f_{SAM}	–19	—

The more abrupt attenuation characteristics, the further efficient high-frequency noise removal and effective signal component output. But an increase in the number of component devices results in less cost effective performance.

Figure 10.6 shows filter's frequency characteristics at 8 kHz (f_{SAM}).

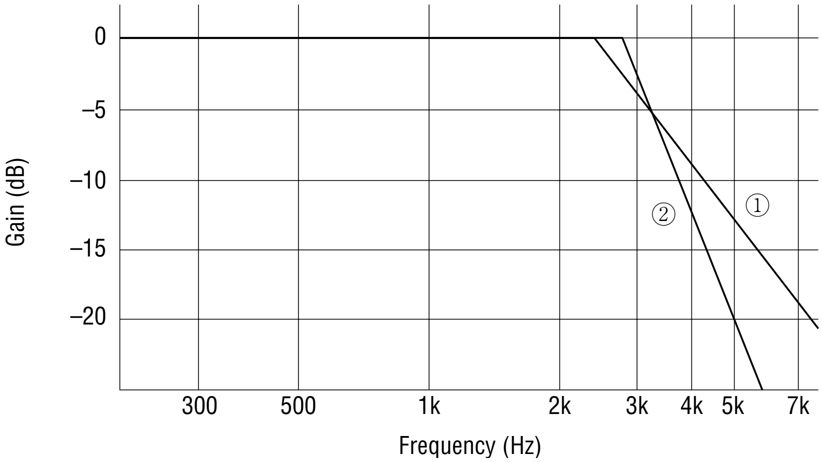


Figure 10.6 Filter's Frequency Characteristics at 8 kHz (f_{SAM})

*1 The cut-off frequency is not strictly defined as the frequency at which attenuation begins. In Butterworth type low-pass filter, the cut-off frequency is defined as the "–3 dB point of total characteristics". In a Chebyshev type low-pass filter, the cut-off frequency is defined as the "first point beyond the maximum ripple amplitude of a passband". Roughly, however, the cut-off frequency can be defined as the frequency at which attenuation begins.

Q7: Is there an inexpensive external filter configuration for voice synthesis?

A7 The active filters (filters using active devices) are classified as Butterworth, Bessel and Chebyshev type filters. These filters can be selected according to the purposes of applications.

The Butterworth type active filter focuses on the flatness of a passband. But characteristics of attenuation and response are inferior to those of the Bessel and Chebyshev type active filters.

For applications where severe flatness of a passband is not required as is the case with LPF used for voice synthesis, the Chebyshev type active filter is recommended where by allowing ripples, abrupt attenuation characteristics can be attained and active filter can be composed by smaller number of parts.

The Chebyshev type active filter can be designed by selecting appropriate ripple amplitude and attenuation characteristics.

If the frequency characteristics of a speaker itself does not reach the desired cut-off frequency, no filter is required.

Configuration

An LPF, consisting of one RC stage, containing no active device is as shown in Figure 10.7, and its transfer characteristics are given by the formula below.

$$F(j\omega) = \frac{\ell_0}{\ell_i} = \frac{1}{1 + j\left(\frac{\omega}{\omega_0}\right)} \quad \text{.........................} \ (1)$$

where

$$\omega_0 = \frac{1}{CR} \quad (\omega_0 = 2\pi f_0)$$

f_0 : Cut-off frequency

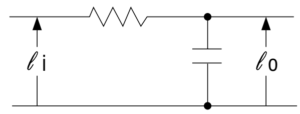


Figure 10.7 LPF Consisting of One RC Stage

Figure 10.8 plots F (jω).

As shown in Figure 10.8, frequency characteristics shows attenuation at –6 dB/oct above ωo. At ωo, a value of –3 dB is observed.

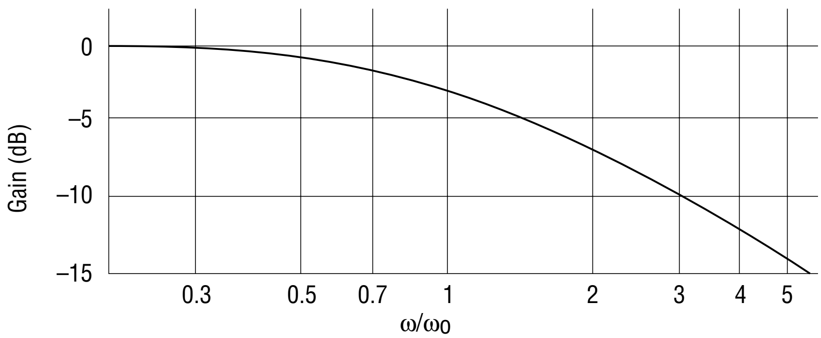


Figure 10.8 Frequency Characteristics of LPF Consisting of One RC Stage

Figure 10.9 illustrates the circuit configuration of a second-order Chebyshev type filter.

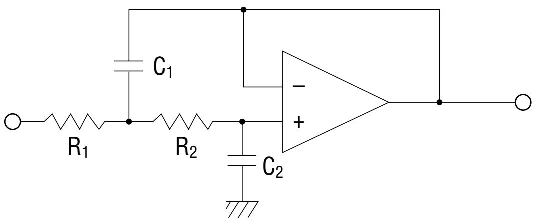


Figure 10.9 Second-order Chebyshev Type Filter

The following gives transfer characteristics of the circuit.

$$F(j\omega) = \frac{\ell_o}{\ell_i} = \frac{1}{1 - \left(\frac{\omega}{\omega_0}\right)^2 + j\,\frac{1}{Q}\cdot\frac{\omega}{\omega_0}}$$

$$Q = \frac{\sqrt{R_1 C_1 R_2 C_2}}{C_2 (R_1 + R_2)} \qquad \omega_0 = \frac{1}{\sqrt{R_1 C_1 R_2 C_2}}$$

The gain is one because the voltage follower consisting of an operational amplifier is used. Assuming that R_1 and R_2 are equal to R, the following expressions are obtained:

$$C_1 = \frac{2Q}{\omega_0 R} \qquad C_2 = \frac{1}{2Q\omega_0 R}$$

Design of high-order Chebyshev type filter

Such even-order filters as the fourth- and sixth-ones can be resolved into second-order elements. Such odd-order filters as third- and fifth-ones can be resolved into second- and first-order (passive filter consisting of one RC stage) elements.

For example, a fourth-order filter can be resolved into two second-order elements as shown in Figure 10.10. Determining fn and qn allows a fourth-order filter to be readily built.

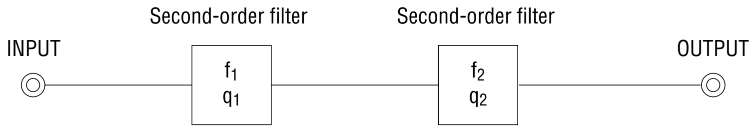


Figure 10.10 Implementation of Fourth-order LPF

The Chebyshev type filter incurs a ripple in the passband. Attenuation characteristics vary with permissible ripples, as fn and qn settings in each stage is changed.

Table 10.4 lists fn and qn values for the Chebyshev type LPF.

Table 10.4 fn and qn Values for Chebyshev Type LPF

	Ripple=0.1dB		Ripple=0.2dB		Ripple=0.25dB		Ripple=0.3dB		Ripple=0.5dB	
	fn	qn	fn	qn	fn	qn	fn	qn	fn	qn
Second order	1.8204497	0.7673593	1.5351966	0.7966418	1.4539722	0.8092536	1.3911667	0.8210811	1.231418	0.8637210
Third order	1.2999029	1.3409276	1.1889612	1.4595033	1.1569921	1.5080264	1.1321861	1.5524768	1.0688535	1.7061895
	0.9694057	0.5*	0.8146341	0.5*	0.7672227	0.5*	0.7292773	0.5*	0.6254565	0.5*
Fourth order	1.1532699	2.1829303	1.0948338	2.4350125	1.0779389	2.5361100	1.0648159	2.6279020	1.0312704	2.9405542
	0.7892557	0.6188010	0.7011094	0.6458968	0.6744223	0.6572494	0.6532428	0.6677803	0.5970024	0.7051102
Fifth order	1.0931318	3.2820141	1.0570753	3.7068586	1.0466301	3.8756825	1.0385110	4.0283601	1.0177347	4.5449633
	0.7974460	0.9145215	0.7472558	1.0009079	0.7324054	1.0359319	0.7207553	1.0678979	0.6904832	1.1778056
	0.5389143	0.5*	0.4614106	0.5*	0.4369509	0.5*	0.4171291	0.5*	0.3623196	0.5*
Sixth order	1.0627261	4.6329012	1.0382299	5.2689021	1.0311242	5.5204164	1.02555981	5.7474076	1.0114459	6.5128456
	0.8344903	1.3315707	0.8030621	1.4917187	0.7938542	1.5556533	0.7866630	1.6135959	0.7681212	1.8103772
	0.5131875	0.5994600	0.4603216	0.6259511	0.4440628	0.6370268	0.4310754	0.6472924	0.3962290	0.6836390

A value of 0.5* for qn indicates the first-order CR stage.

Example of design

The following gives an example of designing the fifth-order Chebyshev type LPF. A permissible ripple is 0.5 dB. Figure 10.11 provides the circuit.

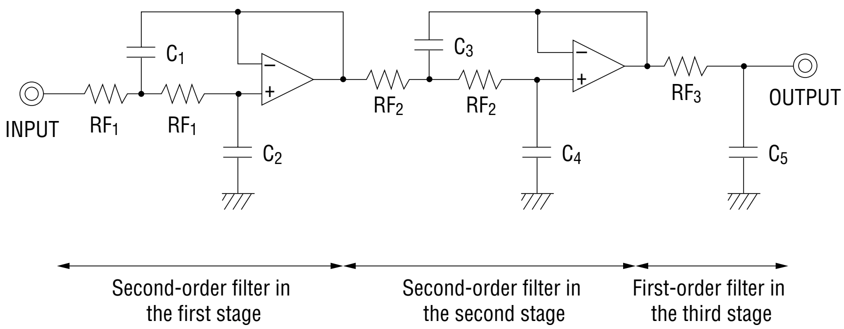


Figure 10.11 Fifth-order Chebyshev Type LPF

The fn and qn values are covered in Table 10.4. Data for the fifth order and a ripple of 0.5 dB provides the following fn and qn values.

For the second-order filter in the first stage fn = 1.0177347 and qn = 4.5449633
For the second-order filter in the second stage fn = 0.6904832 and qn = 1.1778056
For the first-order filter in the third stage fn = 0.3623196 and qn = 0.5

The values above can be used to calculate constants.

When the cut-off frequency is 2.8 kHz, the constants are determined as follows.

Second-order filter in the first stage

R_{F1}=51 kΩ

$$f_0 = 2800\times1.0177347 \fallingdotseq 2850 \text{ (Hz)}$$

$$C_1 = \frac{2Q}{\omega_0 R_F} = \frac{2qn}{2\pi f_0 R_F} = \frac{2\times4.5449633}{2\pi\times2850\times51\times10^3} = 9953 \text{ (pF)}$$

$$C_2 = \frac{1}{2Q\omega_0 R_F} = \frac{1}{2qn2\pi f_0 R_F} = \frac{1}{2\times4.5449633\times2\pi\times2850\times51\times10^3}$$

$$= 120 \text{ (pF)}$$

Second-order filter in the second stage

R_{F2} = 56 kΩ

$$f_0 = 2800 \times 0.6904832 \fallingdotseq 1933 \text{ (Hz)}$$

$$C_3 = \frac{2Q}{\omega_0 R_F} = \frac{2\times1.1778056}{2\pi\times1933\times56\times10^3} = 3463 \text{ (pF)}$$

$$C_4 = \frac{1}{2\pi\omega_0 R_F} = \frac{1}{2\times1.1778056\times2\pi\times1933\times56\times10^3} = 624 \text{ (pF)}$$

First-order filter in the third stage

R_{F3}=68 kΩ

$$f_0 = 2800\times0.3623196 \fallingdotseq 1014 \text{ (Hz)}$$

$$C_5 = \frac{1}{\omega_0 R_F} = \frac{1}{2\pi f_0 R_{F3}} = \frac{1}{2\pi\times1014\times68\times10^3} = 2308 \text{ (pF)}$$

The R_F value can be changed to reflect the actual capacitor value. If C_1 and C_2 values have been multiplied by 1.5, the R_{F1} value should be divided by 1.5.

Selecting appropriate capacitor values leads to the final determination of the filter constants. See Figure 10.12.

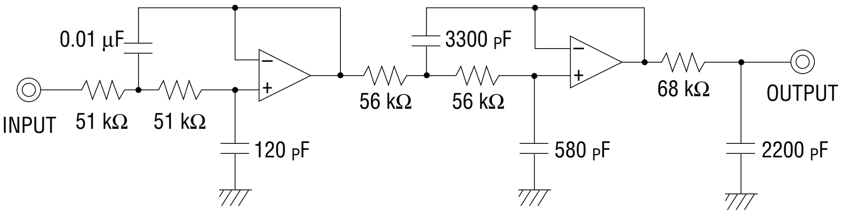


Figure 10.12 Designed Filter (Fifth Order)

Then, filter characteristics in Figure 10.12 are plotted.

Transfer characteristics for the second-order filters in the first and second stages are expressed by the formula below.

$$F(j\omega) = \frac{1}{1-\left(\frac{\omega}{\omega_0}\right)^2 + j\frac{1}{Q}\cdot\frac{\omega}{\omega_0}}$$

Transfer characteristics of the first-order filter in the third stage are expressed by the formula below.

$$F(j\omega) = \frac{1}{1+j\left(\frac{\omega}{\omega_0}\right)}$$

For example, if ω is equal to ωo for the filter in the first stage, we get the following expression.

$$F(j\omega_0) = \frac{1}{1-(1)^2 + j\frac{1}{Q}\times 1} = -jQ = -4.5449633j$$

The following gives the absolute dB value of the I/O voltage ratio.

$$20\ \log\ 4.545 = 13.15\ (\text{dB})$$

The absolute value is 13.15 dB for a frequency of 2850 Hz.

Figure 10.13 provides the plotted dashed curve. The alternate dot-dash line and the alternate two dot-dash line cover the second and third stages, respectively. The solid line provides total characteristics.

As the capacitor values have been approximated, total characteristics indicate that the maximum ripple is 0.6 dB and the cut-off frequency is about 2.9 kHz.

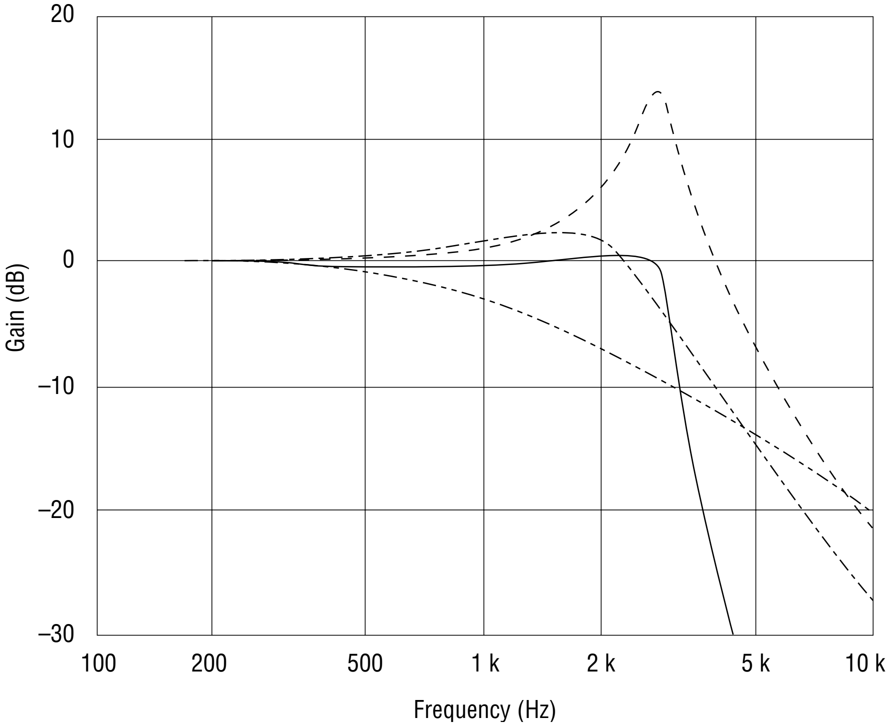


Figure 10.13 Frequency Characteristics of Designed Filter (Fifth Order)

Figures 10.14 and 10.15 provide constants and frequency characteristics of a designed third-order Chebyshev type LPF.

The solid line in Figure 10.15 provides total characteristics.

As the capacitor values have been approximated, the characteristics indicate that the maximum ripple is 3 dB and the cut-off frequency is 2.56 kHz.

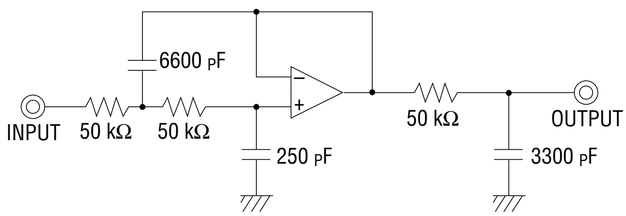


Figure 10.14 Designed Filter (Third Order)

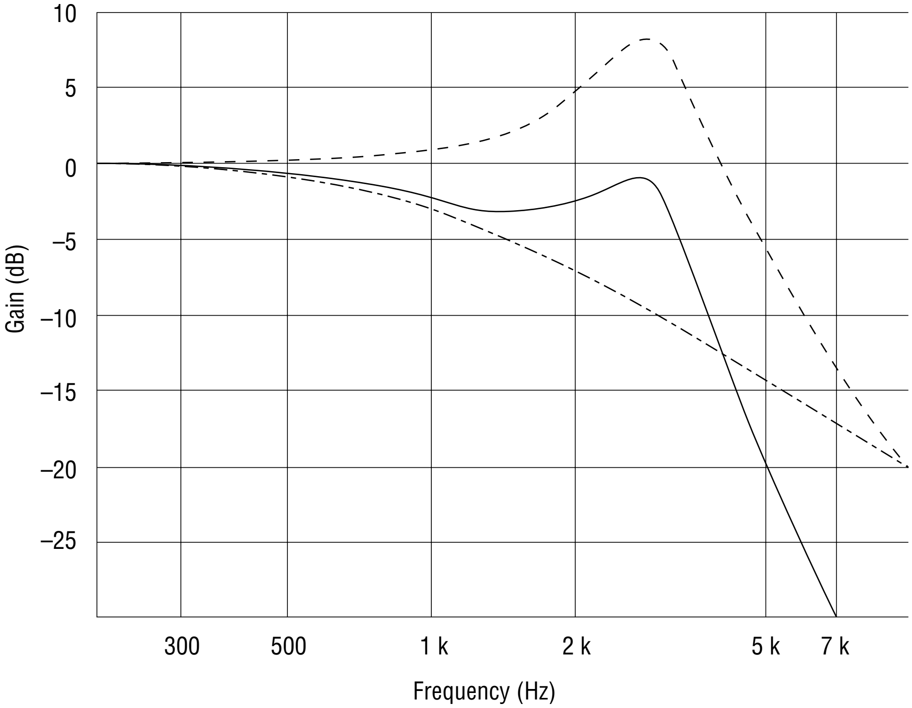


Figure 10.15 Frequency Characteristics of Designed Filter (Third Order)

Q8: Which is the chip's substrate electric potential, the V_{DD} or the GND level?

A8 Table 10.5 lists voice LSI chip products available from OKI and their substrate potential.

Table 10.5 Chip products and their substrate potential

LSI chip product model	Chip's substrate pontential
MSM6658A-XXX	GND
MSM6656A-XXX	GND
MSM6655A-XXX	GND
MSM6654A-XXX	GND
MSM6653A-XXX	GND
MSM6652A-XXX	GND
MSM6650	GND
MSM9805-XXX	GND
MSM9803-XXX	GND
MSM9802-XXX	GND
MSM9836-XXX	GND
MSM6376	GND
MSM6379	GND
MSM6378A	GND
MSM6588	V_{DD}
MSM6722	V_{DD}
MSA180	GND
MSC1157	GND

Q9: How should multiple power supply pins (AV_{DD}, DV_{DD} and so forth) be connected?

A9 For ICs such as MSM6588 having multiple supply and GND pins including AV_{DD}, DV_{DD}, AGND and DGND. The same power supply should be used as shown in Figure 10.16. Connect AV_{DD}, DV_{DD}, AGND, and DGND separately to each V_{DD} and GND on the printed board wiring pattern.

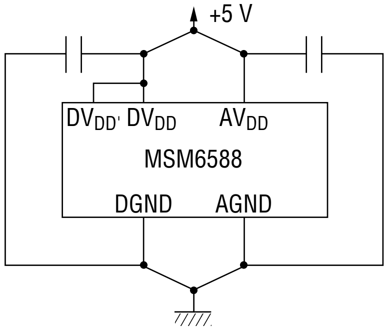
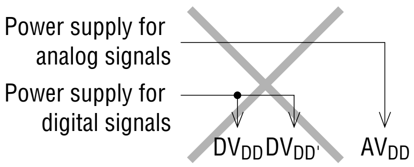
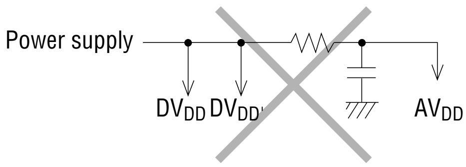


Figure 10.16 Typical GND Connection for MSM6588 Power Supply

The following connection should not be made.





The separation of supply and GND pins for AV_{DD} (analog system) and DV_{DD} (digital logic system) contributes to sound quality enhancement. Ideally a stable potential from separate power sources should be supplied. But in reality supply from separate power sources results in potential differences in the analog and digital systems. Since latching may damage the IC, the same power supply must be used.

Q10: Which EPROM can be used for the MSM6650 and MSM6376 evaluation boards?

A10 Table 10.6 lists the EPROMs (including OTP ROMs) that can be applied to evaluation boards.

Table 10.6 EPROMs Applicable to Evaluation Boards

MSM6650 Evaluation Board	4 Mbits	M5M27C401, MBM27C4001, MSM27C401 (OTP), TC574000, μPD27C4001, and devices with the same pin layout
MSM6650 Evaluation Board MSM6376 Evaluation Board	1 Mbit	M5M27C101, MBM27C1001, MSM271000, TC571000, μPD27C1001, and devices with the same pin layout

Q11: Can voice be output over lines?

A11 Figure 10.17 covers the MSM6650 family (LPF output).

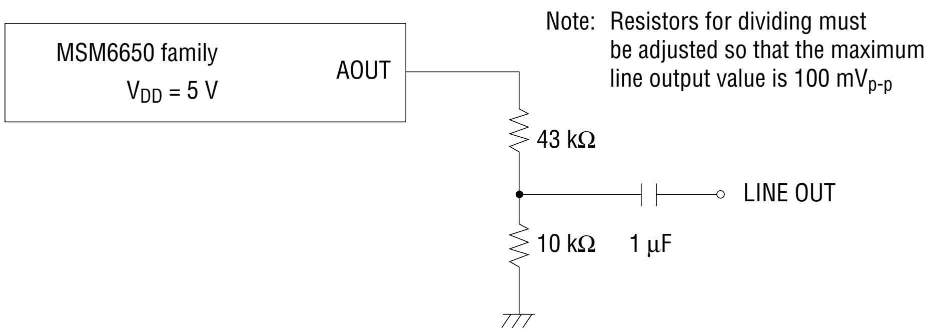


Figure 10.17 Typical Connection for Line Output in MSM6650 Family

Q12: Is voice quality adversely affected by deviation of master oscillation frequency?

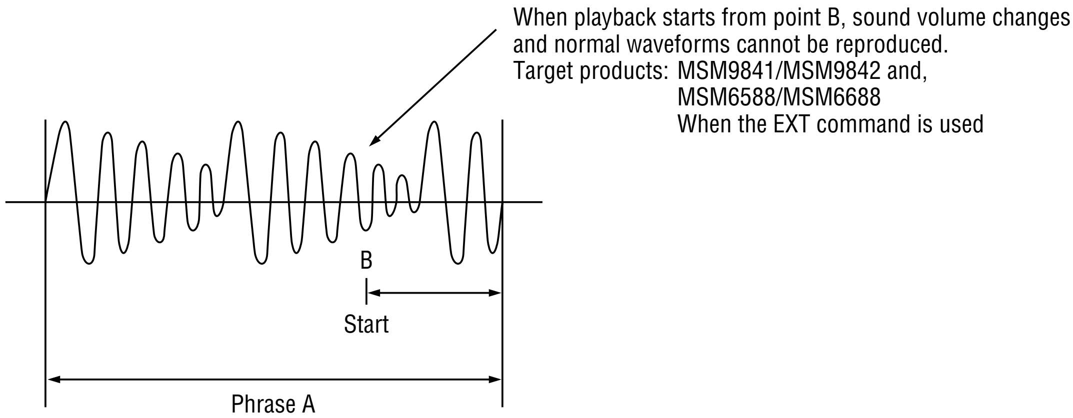
A12 If the master oscillation frequency is within the range of the recommended operating conditions (quaranteed range), the IC will not malfunction.
However, the speech speed and pitch may change with the deviation from the typical value. It is possible to correct the deviation with a voice analysis editing tool before performing voice analysis.

Q13: What kind of recording tape should we use for recording original sound when asking for voice analysis?

A13 Send one of the following recording tapes with original sound.
* Open Reel Tape (tape speed 19cm/sec, normal winding) or
* DAT (Digital Audio Tape)
Do not use a commercialized cassette tape or MD (Mini Disk) because it may adversely affect the quality of the analyzed sound.

Q14: Why does sound volume change when playback starts from part of a phrase?

A14 When playback starts from part of a phrase, normal waveforms cannot be reproduced because the ADPCM algorithm is employed.



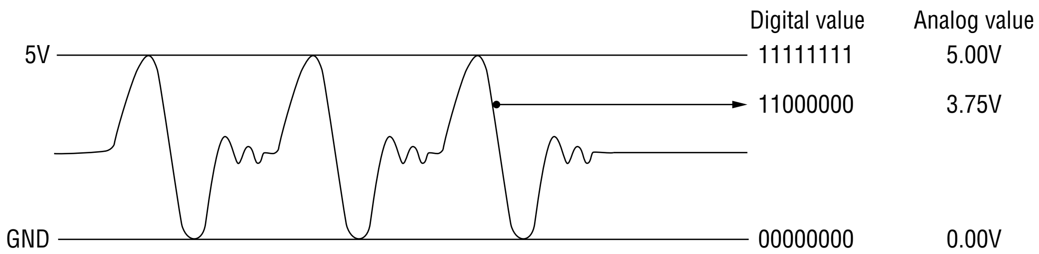
Q15: How can normal waveforms be reproduced when playback starts from part of a phrase?

A15 Do recording/playback continuously without interruption after dividing a phrase to be recorded into many phrases with a recording/playback time of 1 second or 0.5 second.
The reproduced sound is useful though voices between the divided phrases may be missing and its sound quality may be a little bit degraded.
The quality of the reproduced sound should be evaluated before practical use using's demonstration board.
Target products: MSM6588/MSM6688 When the EXT command is used
MSM9841/MSM9842

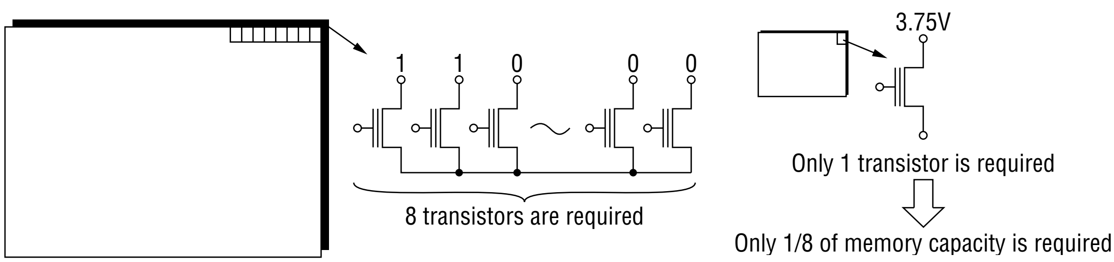
Q16: What is analog flash memory?

A16 The priciple of analog flash memory is described below by comparing digital flash memory and analog flash memory.
In the following waveform, the digital values and analog values are shown, which are sampled when analog signals are input to the 8-bit A/D converter.
When the sampled digital value is written in digital flash memory, 8-bit memory is required. On the other hand, when the analog flash memory is used, only 1/8 of the memory capacity needed for digital flash memory is required because the analog value itself is stored in a memory cell.
The analog flash memory is an innovative memory, which eliminates the need of A/D and D/A converters and requires a small memory capacity because analog signals are input and the analog value is directly stored in a memory cell.

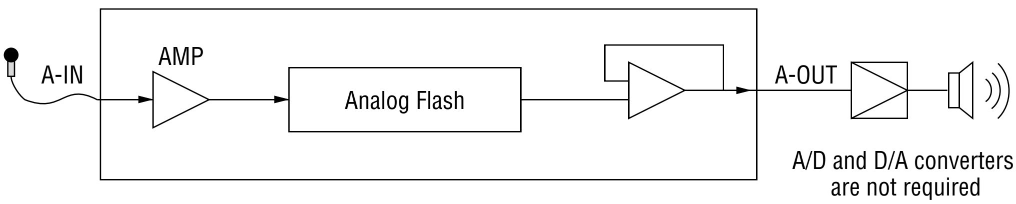
Principle of analog flash memory
Difference between digital flash memory and analog flash memory



When digital flash memory is used **When analog flash memory is used**



Conceptual diagram of recording and playback IC with analog flash memory



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