#### INTRODUCTION

KS16120B is a digital signal processor IC that implements all the functions and hardware interfaces necessary for voice compression, storage and digital telephone answerer.

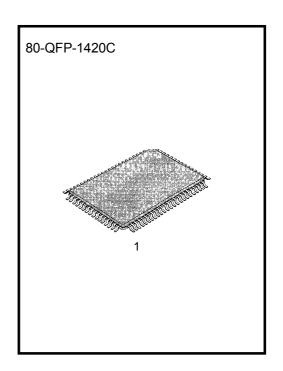
The basic functions include low bit rate speech compression, efficient ARAM management through parity checks and table look-ups, DTMF tone generation and detection, call progress tone detection and high quality voice prompts.

The on-chip interface units provide the access to 4M/16M ARAM, ROM/EPROM and two PCM codecs without any glue logic.

All the clock and control signals, including ARAM refresh, are generated on-chip. KS16120B supports a simple command / status interface protocol for an external host controller. The host writes commands to activate various modes of operations supported by the DSP and read status words to monitor its operation.

KS16120B is manufactured with SAMSUNG 0.8 µ CMOS technology that guarantees reliable performance with low power dissipation.

KS16120B is packaged in a 80-pin plastic QFP.



#### ORDERING INFORMATION

Device	Package	Operating Temperature					
++KS16120BQ	80 - QFP - 1420C	- 20 ~ + 70 ℃					

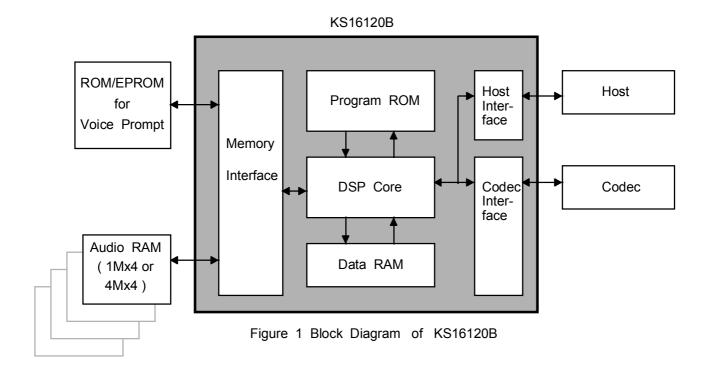
++ Under development

# **FEATURES**

- · High-performance speech compression algorithm
- · Voice activation and silence compaction for longer recording time
- · Advanced audio-grade RAM (ARAM) management for maximum recording time and reliable data storage
- · DTMF detection with near-end echo cancellation, and programmable tone generation
- · Programmable call progress tone detection for busy and /or dial tones
- · Supports high-quality voice prompts from ROM or EPROM up to 64Kbytes
- · Supports multiple message attributes for time stamp, mail box and other applications
- Storage for 128 voice messages and 128 16-bit data or 32 telephone numbers
- 8-bit host interface
- · Supports up to four 1Mx4 or one 4Mx4 ARAMs, refreshed during power-down operation
- 80-pin PQFP package



#### **BLOCK DIAGRAM**

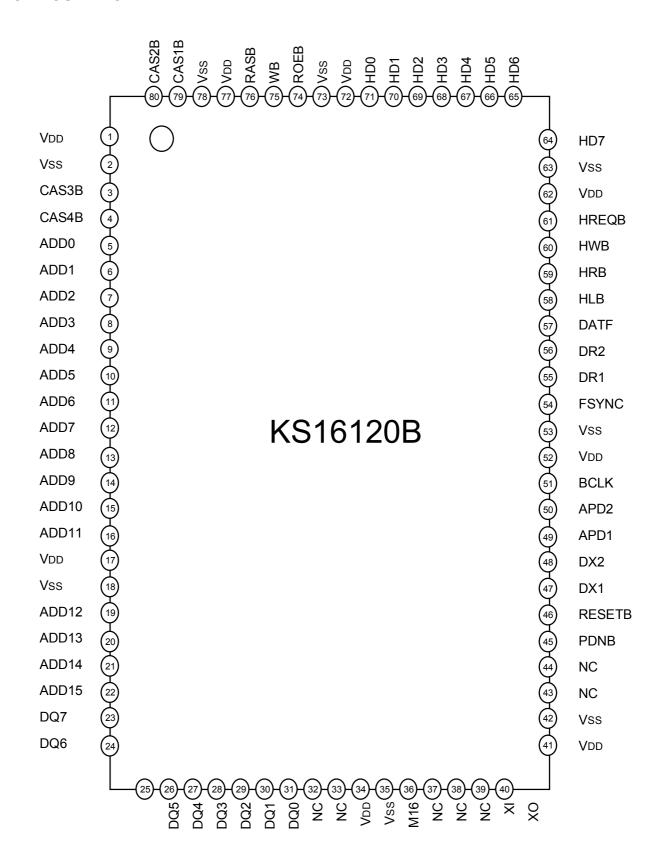


## **CHIP CONFIGURATION**

- KS16120B
  - : SAMSUNG DSP for Digital Answering Phone with ARAM Interface (80 QFP)
- KS8620 / KT8554
  - : Analog in / out interface ( $\mu$ -law PCM CODEC ) -- 16DIP / 16WIDE SOP
- SA5040A
  - : SAMSUNG ARAM with 1M x 4 (4M) or 4M x 4 (16M) organization (20 SOJ)
- External ROM / EPROM
  - : Each 64K-bytes block can supports up to 98.5 seconds of Voice prompts



# **PIN CONFIGURATION**





# PIN DESCRIPTION

Pin Name	Pin	Туре	DESCRIPTION
HD[7:0]	64 - 71	I/O	Host data bus for host instructions and status words from the KS16120B. Pull-up.
HWB	60	ı	Host Write Strobe. A low-to-high transition loads an instruction to the KS16120B.
HRB	59	I	Host Read Strobe. The KS16120B writes a status word to the host data bus.
HLB	58	I	Lower Byte Select. 16-bit command and status words are written or read 8 bits at a
			time. When low, this signal indicates that lower byte is selected.
HREQB	61	0	Host Read Request Indicates that a status word is ready for the host to read. Active
			Low. Goes inactive when the higher byte of a status word is read by the host.
DATF	57	0	Not used.
BCLK	51	0	PCM Data Receive/Transmit Bit Clock for Codec 1 and 2. (2.048MHz clock)
FSYNC	54	0	PCM Data Receive/Transmit Frame Sync for Codec 1 and 2.
APD1	49	0	Codec 1 Inactive Flag. When set, it indicates the codec is not used and may be
			powered down.
DX1	47	0	PCM data Transmit pin to Codec 1. Serial data output from KS16120B to codec.
DR1	55	- 1	PCM Data Receive pin from Codec 1. Serial data output from codec to KS16120B. Pull-up.
APD2	50	0	Codec 2 inactive Flag. When set, it indicates the codec is not used and may be
			powered down.
DX2	48	0	PCM data transmit pin to Codec 2. Serial data output from KS16120B to codec.
DR2	56	I	PCM data Receive pin from Codec 2. Serial data output from codec to KS16120B. Pull-up.
ADD[15:0]	22-19,16-5	0	Address Bus for ARAMs and ROM/EPROM. 10 or 11 LSBs are used for 4M or 16M
			ARAMs, respectively.
ROEB	74	0	ROM Output Enable. Active Low.
RASB	76	0	ARAM Row Address Strobe. Active Low.
CAS1B	79	0	Column Address Strobe for first ARAM. Active Low.
CAS2B	80	0	Column Address Strobe for second ARAM. Active Low.
CAS3B	3	0	Column Address Strobe for third ARAM. Active Low.
CAS4B	4	0	Column Address Strobe for fourth ARAM. Active Low.
WB	75	0	ARAM Read/Write Control. Low for write cycle.
M16	35	I	ARAM Type (4/16M) Select. Indicates 4M when High, 16M otherwise. Pull-up.
DQ[7:0]	23 - 30	I/O	Data bus for ARAMs and ROM/EPROM. 4 LSBs are used for ARAMs. Pull-up.
ΧI	39	- 1	Crystal Input Pin. 24.576MHz.
ХО	40	0	Crystal Output Pin.
RESETB	46	I	System Reset. Active Low.
PDNB	45	I	System Power Down. Active Low.
NC	44,43	I	No connection. Pull-up.
NC	31	I	No connection. Pull-up.
NC	32	I	No connection. Pull-up.
NC	36,37,38	I	No connection. Pull-up.
VDD	1,17,33,41,52	1	Chip Power Supply (5V)
	62,72,77		
VSS	2,18,34,42,53,	I	Chip Ground.
	63,73,78		



# ABSOLUTE MAXIMUM RATINGS (Ta = 25 $^{\circ}$ C)

Characteristics	Symbol	Value	Unit
Supply Voltage	$V_{DD}$	7	V
Input Voltage	$V_{IN}$	$V_{SS}$ - 0.5 to $V_{DD}$ + 0.5	V
Output Voltage	$V_{G}$	$V_{SS}$ -0.5 to $V_{DD}$ + 0.5	V
Storage Temperature	T <sub>STG</sub>	- 65 to +150	${\mathbb C}$

# RECOMMENDED OPERATING CONDITIONS

Characteristics	Symbol	Min.	Тур.	Max.	Unit
Power Supply Voltage	$V_{DD}$	4.5	5	5.5	V
Ground Voltage	V <sub>SS</sub>	-	0	-	V
Operating Temperature	T <sub>OPR</sub>	0	-	70	$^{\circ}$ C
Crystal Frequency	F <sub>CK</sub>	-	24.576	-	MHz
High-level input Voltage	V <sub>IH</sub>	3	-	-	V
Low-level input Voltage	V <sub>IL</sub>	-	-	0.7	V
Current with high-level output	I <sub>OH</sub>	-	-	+1	mA
Current with low-level output	l <sub>OL</sub>	-	-	-1	mA

# DC ELECTRICAL CHARACTERISTICS (VDD = 5V, Ta = 25 ℃, unless otherwise specified)

Characteristics	Symbol	Condition	Min.	Тур.	Max.	Unit
High-level output voltage	$V_{OH}$	I <sub>OH</sub> =100 <i>μ</i> A	VDD - 1.5	-	-	٧
Low-level output voltage	V <sub>OL</sub>	I <sub>OL</sub> =500μA	-	-	Vss + 0.5	V
Input leakage current	I <sub>IN</sub>	V <sub>IN</sub> = 5V	-	25	-	μA
Input pin capacitance	C <sub>IN</sub>	-	-	25	-	pF
Pull-Up resistance	R <sub>PU</sub>	-	-	30	-	kΩ
Operating current at V <sub>DD</sub> =5V	I <sub>DD1</sub>	Normal	-	100	150	mA
and f <sub>OSC</sub> =24.576MHz	I <sub>DD2</sub>	Powered down	-	-	10	mA



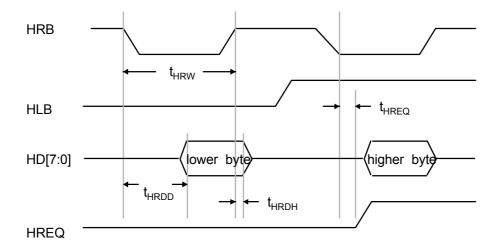
TIMING CHARACTERISTICS ( $V_{DD} = 5V$ , Ta = 25  $^{\circ}$ C , unless otherwise specified )

Characteristics	Symbol	Min.	Тур.	Max.	Unit	Notes
Host read pulse width	t <sub>HRW</sub>	80	- 7		ns	
Host read data delay time	t <sub>HRDD</sub>	00		20	ns	
Host read hold time	t <sub>HRDH</sub>	1			ns	
Host write pulse width	thww	80			ns	
Host write setup time	t <sub>HWDS</sub>	20			ns	
Host write hold time	t <sub>HWDH</sub>	5			ns	
Host read request delay	t <sub>HREQ</sub>			30	ns	
Codec bit clock low time	t <sub>BCKL</sub>		244		ns	
Codec bit clock high time	t <sub>BCKH</sub>		244		ns	
Codec frame sync high delay time	t <sub>BFDR</sub>			10	ns	
Codec frame sync low delay time	t <sub>BFDF</sub>			10	ns	
Codec receive setup time	t <sub>SDB</sub>	30			ns	
Codec receive hold time	t <sub>HBD</sub>	10			ns	
Codec transmit delay time	t <sub>DBD</sub>			30	ns	
ROM address setup time	t <sub>wras</sub>	200	400		ns	
ROM address hold time	t <sub>wrah</sub>	40			ns	
ROM enable pulse width	t <sub>WROE</sub>		400		ns	
ROM data setup time	t <sub>WRDS</sub>	80			ns	
ROM data hold time	t <sub>WRDH</sub>	0			ns	
RASB pulse width	t <sub>RAS</sub>		325		ns	
CASB pulse width	t <sub>CAS</sub>		163		ns	
Row address setup time	t <sub>ASR</sub>		80		ns	
Column address setup time	t <sub>ASC</sub>		80		ns	
Column address hold time	t <sub>CAH</sub>		80		ns	
Access time from CASB	t <sub>CAC</sub>		35		ns	
RASB to CASB delay	t <sub>RAD</sub>	40	80		ns	
RASB cycle time	t <sub>RC</sub>		488		ns	
Write command setup time	t <sub>wcs</sub>	40	80		ns	
Write command hold time	t <sub>wch</sub>	40	160		ns	
WB pulse width	t <sub>wep</sub>		244		ns	
RAM data output setup time	t <sub>DWS</sub>			163	ns	
RAM data output hold time	t <sub>DWH</sub>		163		ns	
CASB setup width (refresh cycle)	t <sub>CSR1</sub>	80			ns	
CASB pulse width (refresh cycle)	t <sub>CAS1</sub>	160			ns	
RASB pulse width ( refresh cycle )	t <sub>RAS1</sub>	160			ns	
Refresh cycle normal operation	t <sub>RC1</sub>		7.8		μs	
Refresh cycle in power-down mode	t <sub>RC2</sub>			16	μs	

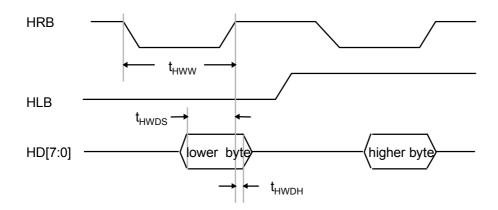


## **TIMING DIAGRAMS**

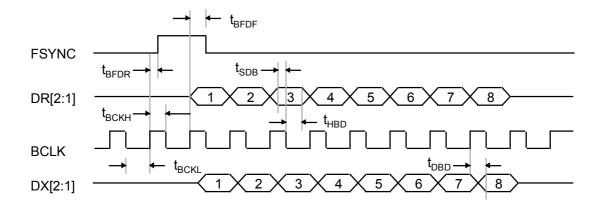
HOST READ CYCLE



# HOST WRITE CYCLE

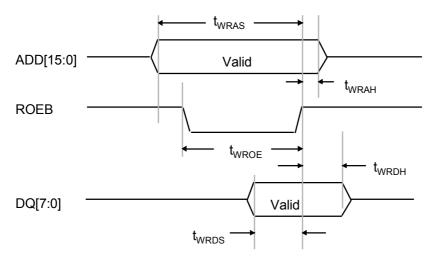


# CODEC READ/WRITE CYCLE

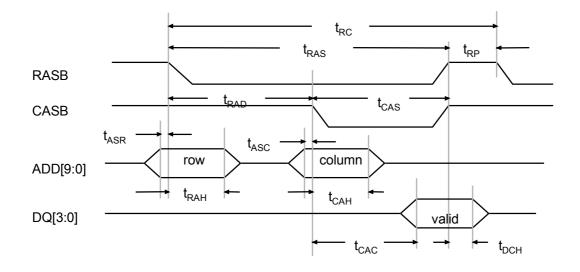




## ROM/EPROM READ CYCLE

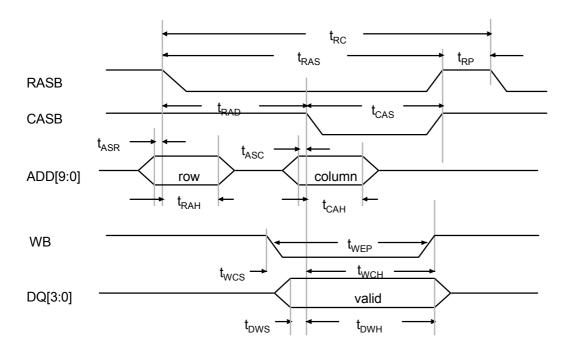


# ARAM READ CYCLE

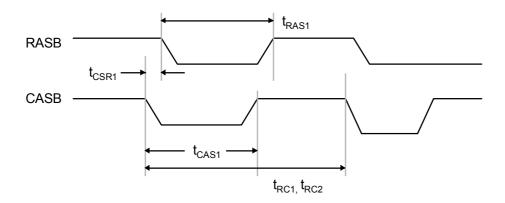




# ARAM WRITE CYCLE



# ARAM REFRESH CYCLE (CAS-Before-RAS)





# HOST INSTRUCTION AND STATUS WORDS

		Instruction																Sta	itus							٦							
MODE		15	14	13	12	11	10	9	8	7	6	5	4	3	2	]	1 0	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
Reset							N	lot	A	ppl	ica	ble						0	0	0	0	WS	0	0	0	0	М	ESS	SAG	Ε (	COL	JNT	
Idle		0	0	0	0	0	0	0	0	0	0		0	0	0	0	0	_	0	0	0	0	0	0	0	0	0	_	0	0	_	-	0
Initialization		0	1	0	1	0	0	0	0	0	С	M	0	0	0	R	AM	0	1	0	1	0	0	0	ES	0	0	0	МВ	R	AM	10	
Record	R0	0	0	0	1	0	0	0	0	SC	VA	LB	0	0	0	0	0 (	0	0	0	1	0	0	0	0	SC	VA	LB	0	0	0	0	0
	R1a	0	0	0	1	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	0	0	0	0	MF	VD	CF	PT	D	TM	F	
	R1b	0	0	0	1	0	0	1		MG	ΑΠ	1	7	ίΟΝ	ΙE	ID	)	0	0	0	1	0	0	1	0	MF	VD	CF	PT	D	TM	F	
	R1c	0	0	0	1	0	0	0	0	0		,	ГАΙ	L (	CUT	Γ		0	0	0	1	0	0	0	0	MF		7	ΓAΙΙ	. C	UT		
Playback	P0	0	0	1	0	0	0		PB	S		N	ИES	SSA	GE	I	D	0	0	1	0	EF	0		PB	Ś		N	MES	SAC	ЭE	ID	
	P1	0	0	1	0	0	0				(	OFF	SE	Γ				0	0	1	0	0	0				(	OFF	SET	1			
	P2a	0	0	1	0	PA	0		PBS	S	0	0	0	0	0	(	0 0	0	0	1	0	PA	0	0	0	EM	0	C	PT	]	DTN	ИF	
	P2b	0	0	1	0	PA	1		PBS	S	0	0	0	0	0	(	0 (	0	0	1	0	PA	1				ÖF	FSE	T				
Voice prompt	V0	0	1	0	0	1	0		PBS			_	RA	_	II	_		0	+	0	0	1	RD	_	<u> </u>	EM	+	_	PT	-	OTN		
	V1	0	1	0	0	0	0	_	PBS		0	_		0	0	Ь	0 0	-	1	0	0	EF	RD	<u> </u>	0	EM	+	-	PT	+-	DTI		
Tone generate		0	1	1	0	0	0	0		MG	ΑΠ	N	7	ΓOΝ	ΙE	ID	)	0	1	1	0	0	0	0	0	0	0	C	PT	]	DTN	ИF	
Message deletion		0	0	1	1	0	0	0	0	0	_	_	SSA	٩GI	ΞΙ	D		0	0 0 1 1 EF 0 0 0 MESSAGE ID														
Garbage collection		0	0	1	1	1	0	0	0	0	0	_	0	0	0	0	_	0	0	1	1	1	0	0	0	-	-	-	0	0	0	0	0
Tone detection		0	1	1	1	0	0	0	0	0	0	_	0	0	0	0	_	0	-	1	1	0	0	0	0	0	Ť	_	PT		DTI		
Read memory state	us	0	1	1	1	0	0	0	1	0	0	Ĺ	0	0	0	0	0	0	1	1	1	RS	0	0		MI	-	1ES	SAC	ъE	CO	UN	Τ
Read data		0	1	1	1	0	0	1	0	0			DA.	ГΑ	II	)					-				DA	TA	_						
Write data	D0	0	1	1	1	0	0	1	1	0			DA	ГΑ	II	)		0	1	1	1	0	0	1	1	0			DAT	ГА	ID	)	
	D1								DA	TA															DA	ATA	1						
Read attribute		0	-	1	1	0	1	0	0	MA		_	ESS	_	ìΕ	ΙΓ		┺							TTI	RIB	UT.	_					
Set attribute	S0	0	1	1	1	0	1	0	1	MA	0	0	0	0	0	(	0 0	0	1	1	1	0	1	_	_	MA	_		0	0	0	0	0
	S1							I	ATT	RH	3U'	ГΕ												1	AT.	ΓRΙ	BU	TE					
Change attribute	U0	0	1	1	1	0	1	1	0	MA		M	ESS	SAC	ìΕ	ΙΓ	)	0	1	1	1	0	1	1	0	MA		Μ	IESS	AG	Е	ID	
	U1					•		1	ΑТΊ	RII	BU'	ГΕ												,	AΤ	ΓRΙ	BU	TE					
Read recording tin	ne	0	1	1	1	1	0	0	0	0	0	0	0	0	0	0	0	0	1	1	1				R	EC	OR	D	TIM	Ε			
H/W select		0	1	0	1	1	0	1	0	0	0	0	0	0	IC	(	OC	0	1	0	1	1	0	1	0	0	0	0	0	0	IC	O	7
Program call	C0	1 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0					1	1 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0							0																		
Progress detector	Ci	COEFFICIENT i											_	DEF	_	_	NT																
Program Tone gen.	Т0	0	1	1	0	1	0	0	0	0	0	0	L.,	TO	NE	Ī	D	0	1	1	0	1	0	0	0	0	0	0	Щ.	TOl	NE	ID	
	T1	0	1	1	0	1	0	0	0	G	ΑI	N 1		G	ΑΠ	N (	0	0	1	1	0	1	0	0	0	(	βAΙ	N 1		G.	AIN	10	
	T2		FREQUENCY 0						L	FREQUENCY 0																							
	T3						]	FRI	EQU	JEN	CY	1											]	FRI	EQU	JEN	VC.	7.1					



# **DESCRIPTION OF INSTRUCTION/STATUS PARAMETERS**

Name	Bit Width	Definition
ATTRIBUTE	16	Message attribute. Each message has two 16-bit attributes that the host can set and
		read for time stamp, mailbox ID and other purposes.
СМ	2	Coding mode. Indicates how much high frequency components of original speech can be
		emphasized . [ 0 : less emphasis , 1 : more emphasis ]
COEFFICIENT i	16	Filter coefficient i for the call progress tone detection filter.
		Used for programming the filter to detect user-defined tones.
CPT	2	Call progress tone detection result. CPT[1] is set when a signal energy is present in
		the tone frequency band. CPT[0] is also set if the signal meets the ON/OFF time
		requirements for a certain period.
DATA	16	16-bit data carrying any information the host chooses, including telephone numbers.
DATA ID	7	Data Index. The KS16120B supports 128 data items.
DTMF	4	Index of the DTMF signal detected 0 for no tone, 1 through 9 for DTMF code 1 through 9,
		and 10, 11 and 12 for DTMF codes , * , 0 and # , respectively.
EF	1	If set, indicates an invalid message index .
EM	1	Indicates the end of message of voice prompt if set, in the PLAYBACK and
		VOICE PROMPT modes, respectively.
ES	1	ARAM memory ( directory area ) test result. When set, it indicates that functional
		faults are detected and the KS16120B can not use the ARAM devices.
FREQUENCY 0	16	Frequency 0. The actual frequency is (8000 / 65536)*(FREQUENCY 0) Hz.
FREQUENCY 1	16	Frequency 1. The actual frequency is (8000/65536)*(FREQUENCY 1)Hz.
GAIN 0	4	Gain of frequency component 0.0xf for -24dB and 0x0 for 6dB in 2dB steps.
GAIN 1	4	Gain of frequency component 1.0xf for -24dB and 0x0 for 6dB in 2dB steps.
GC	1	Garbage collection completion flag. If set, indicates the garbage collection is completed.
IC	1	Input codec selection flag. 0 for codec 1 or 1 for codec 2.
LB	1	Actives the loop back option if set, in the RECORD mode.
MA	1	Message attribute index. Specifies one of the two attributes of a message.
МВ	1	If set, No Area in ARAM is available for ICMs
MESSAGE COUNT	7	The number of messages stored in ARAM.
MESSAGE ID	7	The index of the message to be played - back ( P0) or deleted ( Delete Message )
MGAIN	4	Master tone gain. MGAIN+GAIN0 and MGAIN+GAIN1 should not exceed 15.
MF	1	Memory Full. If set, it indicates no more message can be stored in ARAM.
ос	2	Output codec selection flag. 0 for codec 1, 1 for codec 2 or 3 for both codec 1 and 2.
OFFSET	10	The length of message, from the beginning, to be skipped in unit of seconds.
		It also indicates the current position of message in a status word.



# DSP for Digital Answering phone with ARAM interface

Name	Bit Width	Definition
PA	1	Pause the playback
PBS	3	Playback speed. PBS = 0 or 3 for normal speed speech (NS).
		The speed ranges are from 0.5*NS (twice slower) to 2*NS (twice faster).
		PBS = 1 for 0.5*NS, 2 for 0.75*NS, 4 for 1.25*NS, 5 for 1.5*NS, 6 for 1.75*NS and 7 for 2*NS.
PHRASE ID	7	The index of the phrase in ROM/EPROM to be played - back in VOICE PROMPT mode.
RAM I [3:0]	4	Bit flags for ARAM devices installed. RAM [3] specifies the type of ARAM
		installed ( 0 for 4M or 1 for 16M ). RAM1 [1:0] specifies the total amount of
		memory available ( 0 for 4M, 1 for 8M, 2 for 12M, and 3 for 16M ). RAM1 [2]
		reserved for future use.
RAMO [3:0]	4	Memory test results. Indicates the actual amount of memory available.
		( 0 for 0-1M , 1 for 1-2 M , 2 for 2-3M etc )
RD	1	Ready flag in the VOICE PROMPT mode. If set, it indicates a new phrase may selected.
RECORD TIME	12	Minimum recording time available in unit of seconds.
RS	1	Voice prompt ROM test result. Indicates it passed the test, if cleard.
SC	1	Activates silence compaction if set, in the RECORD mode
TAIL CUT	7	The length of message, from the end, to be removed in unit of 80 msec.
TONE ID	5	Index from the tone table.
		By default, indices 0 through 12 correspond to standard DTMF tones.
VA	1	Enable the voice activation, if set, in the RECORD mode.
VD	1	Voice signal detected in RECORD mode.
WS	1	Indicates a warm start is set, after a reset operation.



#### **FUNCTIONAL DESCRIPTION**

#### HOST INTERFACE

The KS16120B acts as a co-processor to the host. It communicates with the host via 8-bit parallel interface. A simple protocol of 16-bit instruction issued by the host and a 16-bit status word returned from KS16120B is employed.

Through this command / status protocol, the host can directly control the KS16120B to perform multiple functions simultaneously in various modes.

#### MESSAGE RECORDING

The KS16120B uses 4M or 16M ARAMs as the means to store speech data. A recording operation generates a compressed representation of speech segment, or compressed message, and saves it in ARAMs. Subsequent recording operations store the compressed messages sequentially in the memory space is available. A maximum of 128 messages can be stored.

The KS16120B employs a software algorithm running on an on-chip DSP core to compress the in-coming speech samples at 8MHz time intervals from a  $\mu$ -law PCM codec. The algorithm processes the speech on a 20 msec time frame and produces the compressed data at 6.35 kbps.

When used with the silence compaction option, the algorithm can store approximately 14 minutes of compressed speech in a 4M ARAM or one hour of speech in a 16M ARAM.

The silence gaps before a speech signal or between speech segments can be detected and the silence compaction used by the KS16120B to save the memory space needed to store the message. The host may choose the following recording options:

•Voice Activation -- The initial silence is ignored and the recording starts only when the KS16120B detects a speech signal.

Without this option, the recording starts immediately after a record instruction.

•Silence Compaction -- The silence gaps between speech segments are measured and replaced with simulated noise during playback to save the memory space.

A recording operation normally ends with a host command. When necessary, the length of the message may be reduced by removing the tail end of the message, which may contain no speech signal. The instruction from the host specifies the number of speech frames to be removed.



#### MESSAGE PLAYBACK

A playback operation retrieves a compressed message from ARAM. Each message is identified with a message number, given by the host along with a playback command. A portion of memory space is allocated to maintain the information related to individual messages, such as memory address and message length.

The KS16120B reads the compressed message from ARAM and processes it with the decoding algorithm to recover the original speech waveform samples. The samples are fed to an external  $\mu$ -law PCM codec for digital to analog conversion.

The speech of playback is adjustable by the host, from 0.5 to 2 times the normal speed in an increment of 0.25, without changing the voice characteristics.

The host can specify an offset in the playback command. The offset, given in seconds, instructs the KS16120B to skip a specified length of the message, from the beginning, before the playback starts.

During the playback operation, the host can issue a pause command to stop the playback momentarily and resume the operation from the same point.

#### **VOICE PROMPTS**

Certain applications require playing a pre-stored speech segments, including fixed messages, voice prompts and voice digits. The KS16120B has a memory interface that can read these phrases from external ROM or EPROM, without interfering with the messages stored in ARAM.

Voice prompts stored in ROM / EPROM are in compressed form (5.15 kbps) to allow for an increased number of phrases and efficient memory usage. The KS16120B supports a memory space up to 64 Kbytes and supported maximum number of voice prompts is 128. The KS16120B Software Support Tools enable users to generate the ROM / EPROM data for user-specified speech waveform samples.

It is often important to control the time delay or gap between phrases properly when a multiple number of phrases are played in sequence. The KS16120B host interface enables users to control the gap between the current and next prompts accurately in 20 msec resolution.

#### DTMF/CALL PROGRESS TONE DETECTION

During the record, playback and voice prompts modes as well as the tone detection/generation modes, the KS16120B monitors the in-coming signal for the presence of DTMF and call progress tones.

The monitoring result is returned to the host as a part of the 16-bit status word.



# DSP for Digital Answering phone with ARAM interface

The KS16120B employs a near-end echo cancellation ( NEC ) algorithm in its tone detection subsystem. The NEC removes the echo signal, returned by the codec, from the in-coming signal and helps the detector perform reliably in the presence of a strong local signal.

The result of DTMF detection is expressed as a 4-bit number as follows:

TABLE 1. Definition of DTMF Detection Result

Bits	DTMF Code	Freq. 0 ( Hz)	Freq. 1 (Hz)		
0x0	none				
0x1	1	697	1209		
0x2	2	697	1336		
0x3	3	697	1477		
0x4	4	770	1209		
0x5	5	770	1336		
0x6	6	770	1477		
0x7	7	852	1209		
0x8	8	852	1336		
0x9	9	852	1477		
0xa	*	941	1209		
0xb	0	941	1336		
0xc	#	941	1477		

The performance of the DTMF detection is summarized in the following table.

TABLE 2. DTMF Detection Performance

Parameter	MIN.	MAX.	UNIT
Detection signal level	-35	0	dBm
DTMF Twist ( high/ low )		+4/-8	dB
Detection bandwidth		3	%
Noise tolerance (SNR)	12		dB
Tone duration	40		ms
Inter-digit pause	40		ms



# DSP for Digital Answering phone with ARAM interface

In Table 2 and 3, 0 dBm is equivalent to the power of a sinusoid with a peak-to-peak amplitude of 7175 after μ-law to linear conversion. The KS16120B has a programmable call progress tone detector with two output flags. The first one is the continuous tone flag that is set when a signal has a dominant signal energy in the frequency band the user chooses. The second flag indicates whether the input tone meets the ON/OFF interval requirement. During a cold start, the KS16120B sets up the call progress tone detector such that it can detect both the busy and dial tones. The default set-up of the call progress tone detector may be changed by loading a new filter setting through the host interface.

The performance of the default call progress detector is summarized in the following table:

TABLE 3. Performance of Call Progress Detector

Parameter	MIN.	MAX.	UNIT
Detection frequency	330	640	Hz
On/Off Duration	260		msec
Signal level ( detection )	-30		dBm
Signal level ( rejection )		-35	dBm
Noise Tolerance (SNR)	12		dB

#### TONE GENERATION

The KS16120B has a programmable tone generator that send-out a single or dual frequency tone, intended for signalling and other purposes. The total number of tone that can be defined and used is 31. An internal tone table holds the frequencies and gains of individual tones.

The host can instruct the KS16120B to generate a tone signal from an idle or record mode. The KS16120B accepts the 5-bit tone index and 4-bit gain from the instruction. The gain is given in 2-dB setps.

During a system reset, the KS16120B sets the tone frequencies and gains to their default values. The first twelve tones, indices 1 through 12, are standard DTMF tones, specified in Table 1. The host is free to change the frequencies and gains of any tone to meet the need of a specific application.

#### MEMORY/MESSAGE MANAGEMENT

The KS16120B supports a flexible management, from the host, of the messages and other information stored in the memory. The memory space (ARAM and ROM) and the information stored can be manipulated by the host with the following KS16120B features:

•Deletion -- A message is removed from the memory.

The indices of the remaining messages are updated appropriately to reflect the deletion.



- Garbage Collection -- Empty holes in the memory space, created by repeated recordings and deletions, are eliminated to save the space and maximize the recording time.
- •Message Attributes -- Each message stored in memory has two 16-bit attributes to it, which can hold any information relevant to individual message.

The host can write, read or change the attributes and implement the time stamp, mailbox and other applications with them.

•Memory Status -- The host can instructs the KS16120B to report ROM and ARAMs, including the total number of messages stored.

> The host can also instruct the KS16120B to return the approximate time available for recording, computed based on the current memory usage.

•Data Storage -- In addition to the compressed messages, the KS16120B allows the host to store and retrieve 128, 16-bit words in memory. When used for storing telephone numbers, this memory space supports a maximum of 32 numbers, in which each telephone number consists of 4, 16-bit words or 16 hexadecimal digit.

#### POWER DOWN

The KS16120B has an on-chip ARAM refreshing circuitry and maintains the data stored in ARAMs during a power down mode with a minimum power dissipation. When reset after returning from a power down mode, the KS16120B checks the validity of the ARAM data to determine a warm or cold start.

#### AUDIO - GRADE ARAM SUPPORT

The KS16120B helps the system implementation with a support for ARAMs with faulty bits, or Audio-grade ARAM. The KS16120B memory subsystem utilizes an access scheme that uses the mapping of bad blocks, parity check and read-backs, and can cope with time-dependent bit fails. The scheme insures a reliable system operation with the maximum recording capacity.

#### HARDWARE INTERFACE

The KS16120B chip contains a DSP core, program ROM, data SRAM and interface logic. The interface logic includes memory, codec and host interfaces.

## **CLOCKING**

The KS16120B has an on-chip oscillator. It requires a crystal with fc = 24.576MHz.



#### RESET

After the initial system power up, the KS16120B must be reset with a pulse at the RESET pin for a minimum duration of 100 nsec, in order to clear internal registers, set operating parameters to their default values, and initialize the DSP program software. A reset is not required but recommended after a power-down mode.

In the reset mode, the KS16120B first checks the validity of ARAM data and executes either a cold or a warm start. A cold start resets the system, clears any user-defined data and is selected when the ARAM data memory has not been initialized.

A warm start is selected if the KS16120B determines that the ARAM data memory has been properly initialized. In this case, all the user-defined data, including messages, are retained.

#### HOST INTERFACE

The host communicates with the KS16120B through the host interface which is an 8-bit parallel interface with separate read / write strobes, as shown in Figure 2.

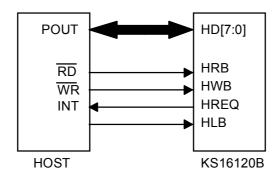


Figure 2. Host Interface for KS16120B

The host writes instructions and reads status words via the 8-bit bi-directional port in two successive accesses. Two bytes of a 16-bit instruction / status word are distinguished with HLB input pin. For proper communication, the lower byte ( with HLB low ) should be accessed first, followed by the higher byte ( with HLB high )

The KS16120B executes the instruction after the higher byte is written by the host. When a status word is ready for the host, the KS16120B pulls the HREQ output low, which may be used as a host interrupt. The host then reads the lower byte of the status, followed by the higher byte at which time the KS16120B sets HREQ high.



In principle, the host issues an instruction in order to get a new status word. The KS16120B returns a status word in response to a command, regardless of whether the previous status output has been read. The response time from an instruction issued and a status returned may be within 20 msec in most cases, except the initialize and garbage collection instructions which return status words after completion of the instructions.

#### CODEC INTERFACE

The KS16120B supports two PCM codecs. However only one codec (DX1 and DR1 ) is used for message recording and playback. The second codec is reserved for full-duplex speaker phone. The codec interface is used for transferring digitized speech samples between the KS16120B and the codec.

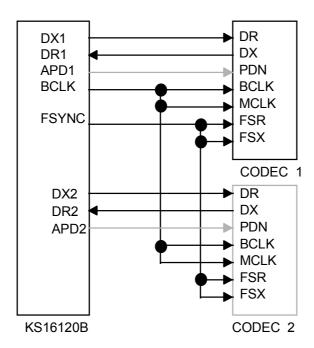


Figure 3. Codec Interface

All the clock and control signals needed to operate the codec are generated by the KS16120B. A linear to μ-law conversion is performed before the KS16120B sends data to the codec and an inverse conversion for the data received.

Each 8-bit data is transferred serially, the sign bit first, through pins DX1 or DR1. The KS16120B generates BLCK and FSYNC to synchronize the transfer.

Output pins, APD1 and APD2, indicate whether the codecs are actively used. These outputs may be used to power down the codecs, but a care should be taken since some codecs may create clicking noise when powered down/up frequently.



#### MEMORY INTERFACE

The KS16120B memory interface subsystem supports connection to ARAMs and ROM / EPROM without any external glue logic. It has a 16-bit address bus (ADD) and 8-bit bi - directional data bus (DQ), shared by ARAMs and ROM/EPROM.

Output pin ROE should be used as an output enable for the ROM/EPROM.

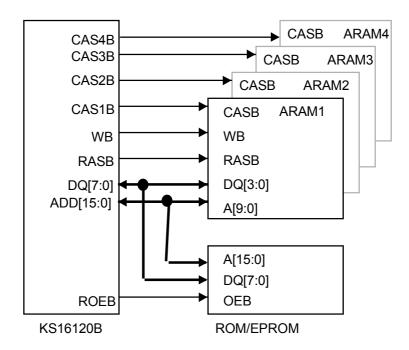


Figure 4. Memory Interface with 4M ARAMs

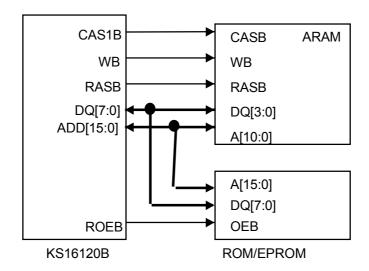


Figure 5. Memory Interface with 16M ARAMs



The KS16120B supports both 4M and 16M ARAMs in a by-4 arrangement. It can be use up to four 4M ARAM devices for the message memorized space. 4 lower bits of the data bus (DQ) are used for transferring data while the lower 10 or 11 bits address bus (ADD) are used for column and row addresses of 4M and 16M devices, respectively. The memory interface subsystem has an on-chip ARAM refreshing control ( CAS - before - RAS ) and bit -error checking circuitry. The refresh cycle is 7.8 mSec in normal operation and 15.6 mSec during the power down. The data transfer from the ROM / EPROM is byte - wide with the maximum address space of 64 K bytes.

#### SOFTWARE OPERATION

After a system reset, the KS16120B is in aidle mode. The host has to issue a new command to put it in active mode. The KS16120B interprets the command, performs the functions specified by the command, and returns a status word. Section HOST INSTRUCTIONS AND STATUS WORDs summarizes the instruction set supported by the KS16120B and the status words returned for individual instructions.

The KS16120B software operational modes with corresponding instructions and status words are described in this section. In each mode of operation, the host is expected to issue a sequence of valid commands in an appropriate order. The KS16120B returns to the IDLE mode when a task terminates or an invalid command is detected, except where described otherwise in this section.

#### RESET

The KS16120B enters a mode with an input pulse at the RESET pin. The reset is required after initial system power-up. It is not required but recommended after a period of power-down operation.

The reset operation first determines whether a cold start or a warm start is needed by checking the validity of ARAM data. If the initialization has been performed before a power-down, the ARAM data be valid, resulting in a warm start. A cold start is performed otherwise. A cold start initializes all system variables, user-defined parameters and memory space.

On the other hand, a warm start retains the messages stored and other user-defined parameters.

At the completion of the start - up procedure, the KS16120B returns to the IDLE mode with the status word indicating a warm or cold start and the number of messages stored.

#### **IDLE**

The KS16120B monitors the host interface for a new instruction in this mode. A new task may be initiated from this mode. It returns to the IDLE mode at the completion of a task automatically or by an explicit idle command from the host.



#### INITIALIZATION

When the host issues an initialize instruction, the KS16120B initializes the message memory space (ARAMs) and returns the result to the host. The host specifies the total amount of memory space available in parameter field, RAMI, in unit of 4M bits while the type of ARAM is selected with the M16 input pin (4M or 16M).

The KS16120B checks the available memory space exhaustively for faulty bits, creating a table of usable blocks which is used for memory access during normal operation. The time it takes for the KS16120B to complete the initialization depends on the memory size and is approximately 3 sec / 4Mbits.

The KS16120B returns to the IDLE mode at the completion.

The status field RAMO returns the actual size of memory for recording in unit of 1M bits.

#### **RECORD**

The message recording starts with the R0 instruction.

The host may specify the following options with the instruction:

- •Voice Activation (VA) -- The recording starts only after voice signal is detected.

  The KS16120B starts recording immediately without this option.
- •Silence Compation (SC) -- The silence gaps during a record are detected and compacted to save the memory.
- •Loop Back (LB) -- The in-coming voice signal is looped back to the codec input.

Once the KS16120B is in the RECORD mode, the host may issue the following instructions to the KS16120B

- •R1a -- monitoring the recording status (MF,VD) and the DTMF and all call progress tone detection result (DTMF, CPT).
- •R1b -- generate a tone, in addition to the tasks performed with R1a. See the TONE GENERATION mode for definition of the input parameters. The tone generation stops with another R1b instruction with TONE ID = 0.



See TONE DETECTION mode for a description of the DTMF and call progress detection result. The recording status is reported with the status bits, MF and VD, which indicate the memory full and detected voice conditions, respectively.

The KS16120B continues the recording which it reads and responds to these commands. If needed, the host can use these instructions repeatedly without interrupting a recording operation. When the memory becomes full, the KS16120B remains in the RECORD mode even through the message is not stored any more.

The host may terminate a recording with an idle instruction. The message is saved in the memory before the KS16120B returns to the IDLE mode. Alternatively, the host may issue an R1c instruction to terminate recording and remove the tail portion of the recorded message.

The 7-bit parameter, TAIL CUT, specifies the length of tail to be removed in the unit 80 msec.

#### **PLAYBACK**

The KS16120B enters the PLAYBACK mode with the P0 instruction, which also specifies the played back speed (PBS) and the index of message to be played (MESSAGE ID). The playback speed ranges is from 0.5 (twice slower) to 2 (twice faster) in the step of 0.25.

The actual playback starts after the P1 instruction is received. The OFFSET bits of the instruction specifies the length of the message, from the beginning, to be skipped in unit of seconds.

While the KS16120B plays a message, the host may issue one of the following instructions:

P2a -- monitor the DTMF, call progress detection result (DTMF, CPT) and recording status (EM). P2b -- read the current position in the message, from the beginning (OFFSET).

The playback continues with these instructions if the PA bit is zero.

The host can pause the playback with either instruction with PA = 1, in which case the KS16120B stops the playback operation and resumes it when a new instruction (P2a, P2b) is entered with PA = 0.

The KS16120B stays in the PLAYBACK mode and continues to respond to the P2a and P2b instructions even when it is in PAUSE (PA = 1) or the end of message is reached (EM = 1). It returns to the IDLE mode with an idle instruction.



#### VOICE PROMPT

The host issues a voice prompt instruction (V0) to play a phrase that is pre-stored in ROM/EPROM. The instruction specifies the phrase number and the desired playback speed.

Once in the VOICE PROMPT mode, the host can issue the V1 instruction to monitor the status of the phrase playback ( RD, EM ) and the DTMF / call progress detection result ( DTMF, CPT ) without interrupting the playback.

The KS16120B sets the status bit, RD (ready), when it reaches near the end of the phrase. A new V0 instruction with an appropriate phrase number may be issued to play another phrase. The host can play two phrases without any gap between them by issuing a V0 instruction immediately after RD is set. Alternatively, the host may create a silence gap between phrases by delaying the instruction. Note that, the KS16120B returns a status word every 20 m sec period. (For instance, a 40-msec gap is created by a command sequence, V1 or V1 and V0, immediately after RD has been set).

The KS16120B sets the status bit EM when it actually reaches the end of a phrase. The VOICE PROMPT mode terminates with an idle instruction from the host.

#### TONE GENERATION

A tone signal is generated and send to the codec in this mode. The tone is selected from the tone table that contains 31 tone definitions. Each tone may have two frequency components with gains, respecitively. Refer to the PROGRAMMING mode for their definitions.

The KS16120B enters the TONE GENERATION mode with a tone generate instruction that specifies the tone index ( TONE ID ) and master gain ( MGAIN ). The actual gains of individual frequency components are determined by MGAIN + GAIN0 and MGAIN + GAIN1. These sums specify the tone gains , ranging from - 24dB to 6dB in 2dB steps and hence must not exceed 15. Here 0dBm is equivalent to the power of a sinusiod with a peak to peak amplitude of 7175 before a linear to  $\mu$ -law conversion.

In the TONE GENERATION mode, the host may switch to a different tone signal by sending a new tone generate commmand with the appropriate TONE ID and MGAIN.

The DTMF and call progress detection result may be monitored, without interrupting the tone generate, by issuing a tone generate command with the same tone parameters.

The TONE GENERATION mode terminates with an idle command.



#### MESSAGE DELETION

In this mode, the KS16120B removes specified message in the MESSAGE ID field of the instruction and returns to the idle mode. The status bit (EF), indicates an invalid message index, if set.

#### GARBAGE COLLECTION

In this mode, the KS16120B eliminates empty holes in the memory space by removing one message at a time. A status word is returned after each remove. When all the messages have been removed, the KS16120B sets the status bit (GC) and returns to the IDLE mode.

The time it takes to complete a garbage collection varies depending on the size of memory and the number of messages stored.

#### TONE DETECTION

This mode monitors the in-coming signal for the presence of DTMF and call progress tones. While in this mode, the KS16120B checks for a new host instruction every 20 msec and returns a status word in response to a new command.

The result of the detection is passed to the host, in the status fields, DTMF and CPT.

#### READ MEMORY STATUS

When a read-memory-status instruction is issued by the host, the KS16120B sends the information on the voice prompt ROM / EPROM and message memory as well as the number of messages stored, through the status word and returns to the IDLE mode.

#### READ DATA

The KS16120B provides a storage of 128 16-bit words that may contain any information including telephone numbers. When the read-data command is issued, the status word contains the data specified in the DATA ID field of the instruction and the KS16120B returns to the IDLE mode.

#### WRITE DATA

The D0 instruction specifies the index of data to stored.

The KS16120B then reads the 16-bit data in D1, stores it and returns to the IDLE mode.



#### READ ATTRIBUTE

When the read-attribute instruction is issued by the host, the KS16120B sends out the 16-bit attribute of the message specified by the MESSAGE ID and MA field of the instruction, and returns to the IDLE mode. Note that each message has two attributes, identified by the MA filed

An invalid message index in the instruction causes the KS16120B to return to the IDLE mode with the status of 0x7000.

#### SET/ CHANGE ATTRIBUTE

An S0 instruction is entered by the host to set a message attribute with MA = 0 or 1. The host then writes the 16-bit attribute in the following S1 instruction. The KS16120B attaches this attribute to the next message to be stored.

The host may change an attribute of an existing message with the change-attribute instructions. The U0 instruction specifies the index of messaage and the type of attribute to be changed, followed by the 16-bit attribute data (U1).

The KS16120B returns to the IDLE mode after the attribute is set or changed, and also when an invalid message index is detected (status = 0x7000).

#### READ RECORDING TIME

In this mode, the KS16120B sends the minimum available recording time and returns to the IDLE mode. The RECORD TIME field of the status word represents the time in unit of seconds.

# **HW SELECT**

The host may select the codecs for voice input for recording and voice output for playback with this command. The input codec is either codec 1 or 2. The output codec may be either codec 1 or 2 also. The default codec is codec 1.



#### PROGRAM CALL PROGRESS DETECTOR

The host may load filter cofficients to be used for the call progress detection in this mode, initiated with the C0 instruction. The subsequent instructions, C1 or C15, set the 15 filter coefficients. The KS16120B returns to the IDLE mode after reading the fifteenth coefficient.

#### PROGRAM TONE GENERATOR

In this mode, the host can change the definition of a tone in the KS16120B tone table. The T0 instruction specifies the tone index ( TONE ID ) and must be followed by T1, T2 and T3, in that sequence.

The T1 instruction defines gains of the two frequency components (GAIN0 and GAIN1) of the tone. The actual gain of each frequency is determined by the sum of GAIN0 or GAIN1 with the master gain (MGAIN) given with the instruction of tone generation.

The range of gains ranges from -24dB (sum = 0xf) to 6dB (sum = 0x0) in 2dB steps. Setting either GAIN1 or GAIN1 to zero results in a single frequency tone.

Instructions T2 and T3 determine the tone frequencies with parameters, FREQUENCY 0 and FREQUENCY 1, respectively. The actual frequency selected is (8000/65536)\*FREQUENCY 0 or 1.

The T3 instruction completes this mode and the KS16120B returns to the IDLE mode.



# 80-QFP-1420C

