

## 1.0 GENERAL DESCRIPTIONS

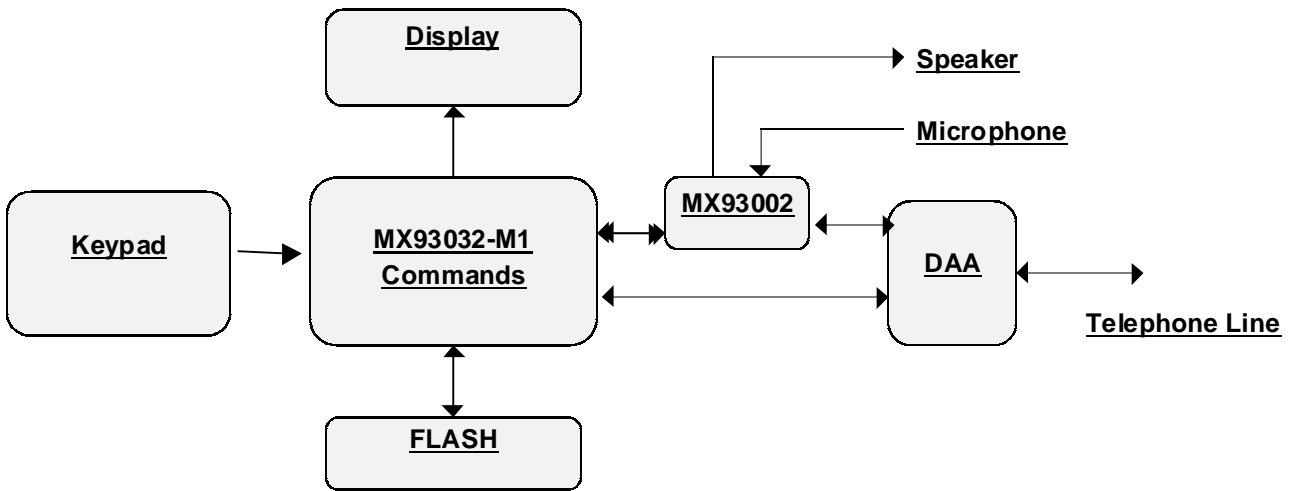
- MX93032-M1 is an engine chip on which digital answering machine (DAM) with full duplex speaker phone is implemented. Besides, CAS tone detection capability and FSK decoder for Caller-ID service has also been implemented as well.
- It provides speech processing / management modules, including speech compression/decompression, silence management, telephone line signal processing, Macronix MX29F16A FLASH management, line echo cancellation and acoustical echo cancellation, etc.
- The MX93032-M1 has built-in DSP mode and MCU mode. In DSP mode, users do not need external microprocessors and can effectively reduce the overall system cost in DAM with full-duplex speakerphone system. In MCU mode, the MX93032-M1 is controlled by an MCU (micro controller) via a 8 bit parallel or serial interface.

## 2.0 FEATURES

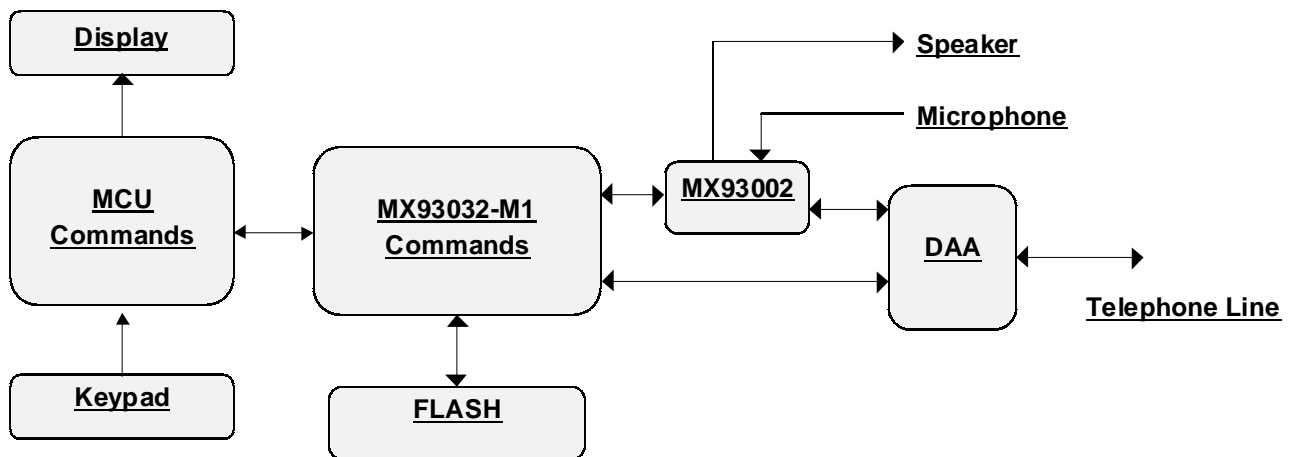
- Switchable 4.8kbps/12.8kbps speech compression/decompression algorithm for message recording/playback.
- Minimum 15 or 40 minutes of recording time for one 16M bits MX29F16A FLASH based on 12.8K or 4.8K compression rate, respectively.
- Low (4.8kbps) or High (12.8kbps) compression rate with silence management technique for voice prompt making. Voice prompt can be stored in Flash memory.
- Support total 127 messages, maximum 7 OGMs and maximum 8 personal mailboxes.
- DTMF generation and detection with near-end echo cancellation.
- Provide single low crystal solution (FLL).
- Built-in DSP mode and MCU mode. In DSP mode, user can mask the system code into the MX93032-M1. In MCU mode, user can control the MX93032-M1 via MCU (micro controller).
- Provide standard sample program written by 8051 assemble for customer development.
- Full duplex speakerphone with both acoustical echo canceller (AEC) and line echo canceller (LEC)
- Built-in CAS signal detector for monitoring the presence CPE alerting signal in Calling Identity Delivery on Call Waiting.
- FSK decoder for Calling Identity Delivery on CID Type I and CID Type II
- Caller ID Type II can be turned on under speaker phone mode
- Auto detecting MCU mode and DSP mode

## 3.0 BLOCK DIAGRAM

### DSP MODE

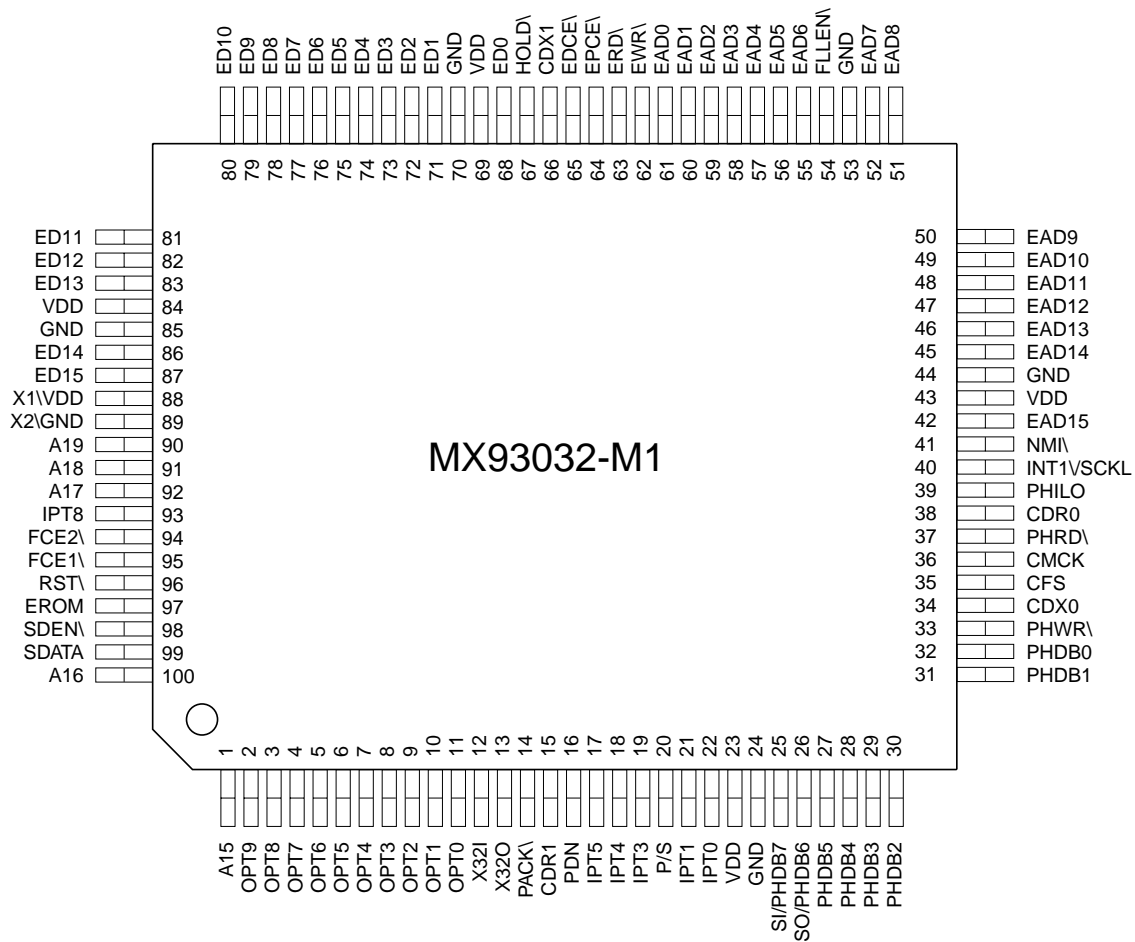


### MCU MODE

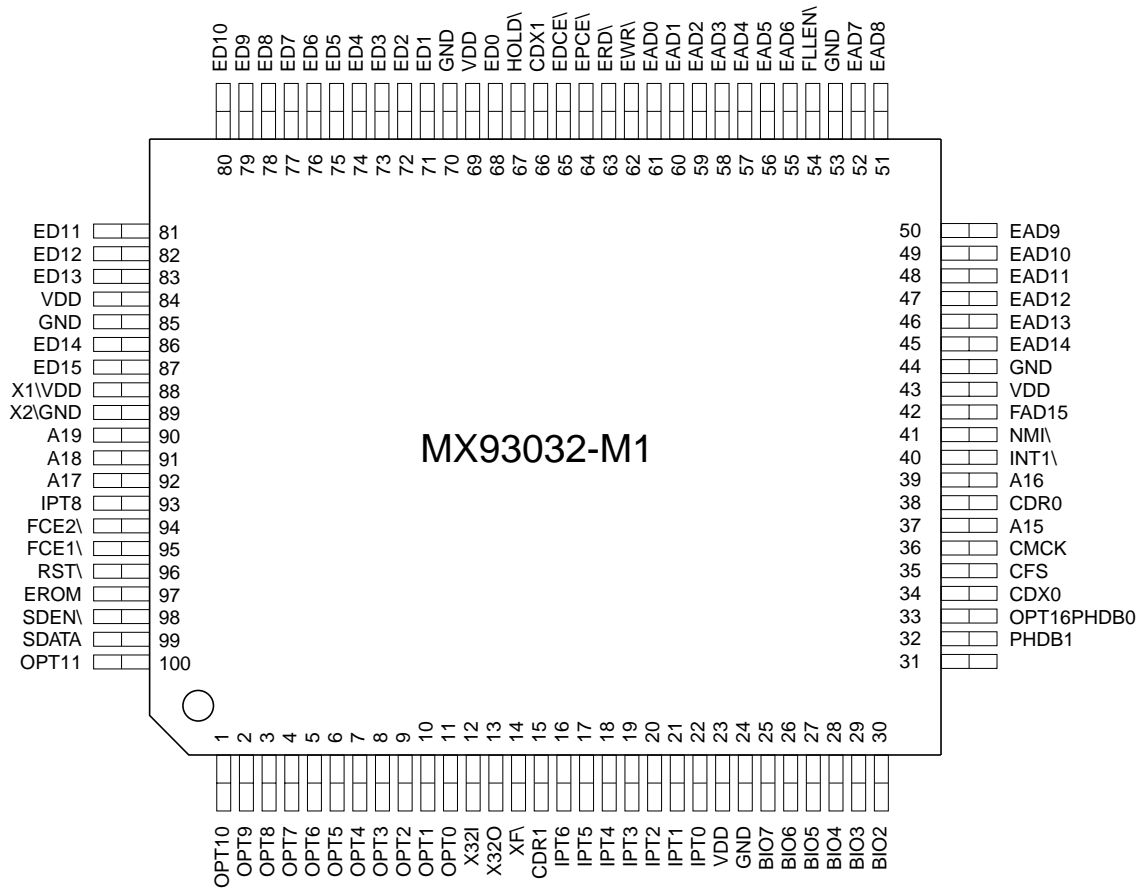


## 4.0 PIN CONFIGURATIONS

### 4.1 MCU MODE



## 4.2 DSP MODE



## 5.0 PIN DESCRIPTIONS

### POWER/CLOCK/CONTROL PINS :

SYMBOL	PIN TYPE	PIN NUMBER	DESCRIPTION
VDD		23, 43, 69, 84	5V power source
GND		24, 44, 53, 70, 85	Ground
X1/VDD		88	32.256MHz crystal input/Connect to VDD in single low X'tal mode
X2/GND		89	32.256MHz crystal output/Connect to GND in single low X'tal mode
RST\	IS	96	Power-on reset
XF\	OA	14	External flag if UPMODX=1. This pin can be directly written by one DSP instruction. Default inactive (5V output)
HOLD\	IS	67	Hold DSP clock down and release bus
EROM	IS	97	Disable internal ROM, use external ROM only
NMI\	IS	41	Non-maskable interrupt pin
INT1\	IS	40	Interrupt pin
X32O		13	32.768KHz crystal output
X32I		12	32.768KHz crystal input
FLLEN\	IS	54	1 : Dual X'tal mode 0 : Single low X'tal mode
P/S	IS	20	1:Parallel interface in MCU mode 0:Serial interface in MCU mode
PDN	IS	16	0:POWER DOWN

### CODEC INTERFACE PINS :

SYMBOL	PIN TYPE	PIN NUMBER	DESCRIPTION
CFS	OA	35	Codec frame sync, 8KHz (9.6KHz). Output low in power down mode.
CMCK	OA	36	Codec master clock, 1.536MHz. Output low in power down mode.
CDX0	OA	34	Codec data transmit
CDR0	IS	38	Codec data receive
CDX1	OA	66	Codec data transmit
CDR1	IS	15	Codec data receive
SDEN\	OB	98	Serial data enable(for MXIC codec family)
SDATA	OB	99	Serial data transmit(for MXIC codec family)

### IPT : Input port

SYMBOL	PIN TYPE	PIN NUMBER	DESCRIPTION
IPT4-IPT5, IPT8	IS	18-17,93	Input port
IPT0-IPT3	ISH	22-19	Input port with internal pull high resistor (R=30KW)

**OPT : Output port**

SYMBOL	PIN TYPE	PIN NUMBER	DESCRIPTION
OPT0-OPT9	OB	2-11	Output to pin, all output values are registered and may be read back when read by "IN" instruction.
OPT10-11	OB	1, 100	Output to pin, all output values are registered and may be read back when read by "IN" instruction. Only available for DSP mode

**MEMORY INTERFACE PINS :**

SYMBOL	PIN TYPE	PIN NUMBER	DESCRIPTION
EAD0-EAD15	OA/Z	61-55, 52-45, 42	External memory address bus.
ED0-ED15	IT/OA/ZR	68, 71-83, 86-87	External memory data bus. With soft latch feed back current is 250 $\mu$ A
EDCE\	OA/Z	65	External data chip enable
EPCE\	OA/Z	64	External program chip enable
ERD\	OA/Z	63	SRAM/ROM/IO external read
EWR\	OA/Z	62	SRAM/ROM/IO external write
A15	OA	37 (DSP MODE) 1 (MCU MODE)	address bus for FLASH
A16	OA	39 (DSP MODE) 100 (MCU MODE)	address bus for FLASH
A17	OA	92	address bus for FLASH
A18	OA	91	Address bus for FLASH
A19	OA	90	Address bus for FLASH
FCE1\, FCE2\	OB	95, 94	FLASH chip enable

**uP Parallel Interface (MCU MODE)**

SYMBOL	PIN TYPE	PIN NUMBER	DESCRIPTION
PHDB0-PHDB7	IS/OA/Z	25-32	Parallel data bus
PHILO	IS/OA/Z	39	Select High or low byte. 1: high byte , 0: low byte
PHRD\	IS/OA/Z	37	HOST read enable
PHWR\	IS/OA/Z	33	HOST write enable
PACK\	OA	14	Response is ready

**uP Serial Interface (MCU MODE)**

SYMBOL	PIN TYPE	PIN NUMBER	DESCRIPTION
SCLK	IS	40	Synchronous clock
SI	OA/Z	25	HOST serial data in
SO	IS/Z	26	HOST Serial data out

**BIO: Bi-directional I/O (DSP MODE)**

SYMBOL	PIN TYPE	PIN NUMBER	DESCRIPTION
BIO7-BIO0	IT/OA	25-32	Input/output port when UPMODX = 1. Direction is controlled by BIO15-BIO8 (see BIOR).

**NOTE :**
**IT** :TTL level input

**IS** :CMOS level Schmitt trigger input (hysteresis : 2V~3V)

**ISH** :CMOS level Schmitt trigger input with internal pull high resistor (~30KW)

**OA** :8mA drive level output

**OB** :16mA drive level output

**Z** :high impedance state

**ZR** :high impedance state with soft latch

**MULTIPLEX PINS:**

PIN NUMBER	PIN NAME	DSP MODE	PIN NAME	MCU MODE
25~32	BIO(7:0)	Input/output port	PHDB(7:0)	Host data bus
39	A16	Flash address	PHILO	High low data select
37	A15	Flash address	PHRB\	Host read
33	OPT16	Output port	PHWR\	Host write
14	XF\	External flag	PACK\	Acknowledge to host
100	OPT11	Output port	A16	Flash address
1	OPT10	Output port	A15	Flash address

PIN NUMBER	PIN NAME	FLEN=1 (Dual X'tal)	PIN NAME	FLEN=0 (Single X'tal)
88	X1	32.256MHz crystal input	VDD	Power VDD
89	X2	32.256MHz crystal output	GND	Power ground

NOTE FLEN: pin 54

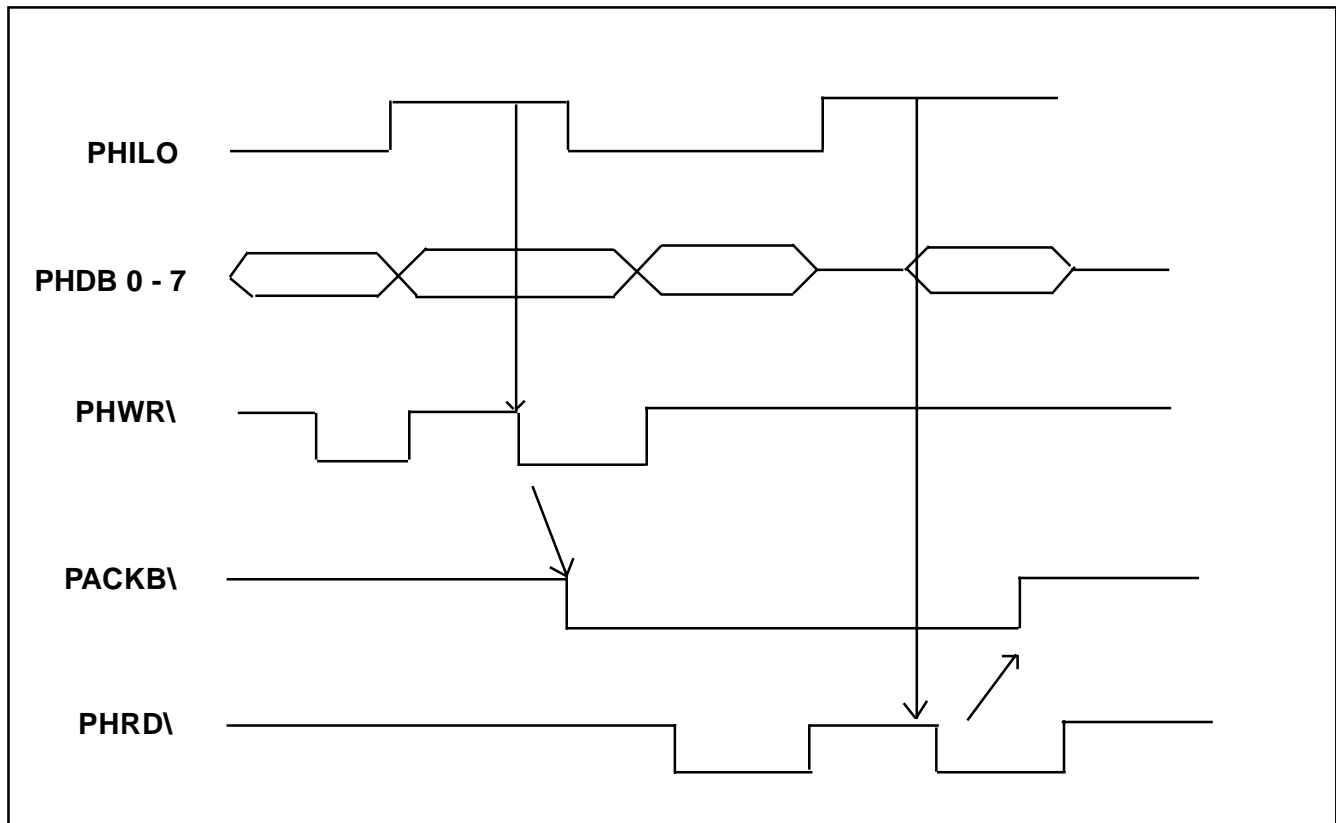
## 6.0 FUNCTIONAL DESCRIPTIONS

### HOST INTERFACE (MCU MODE)

The command and response of the MX93032-M1 are made up by a 16-bit code. The protocol that one command sent and one response received must strictly be obeyed by HOST ; otherwise, dead lock will occur. Meanwhile, both parallel interface and serial interface are provided by the MX93032-M1.

#### 8-bit parallel :

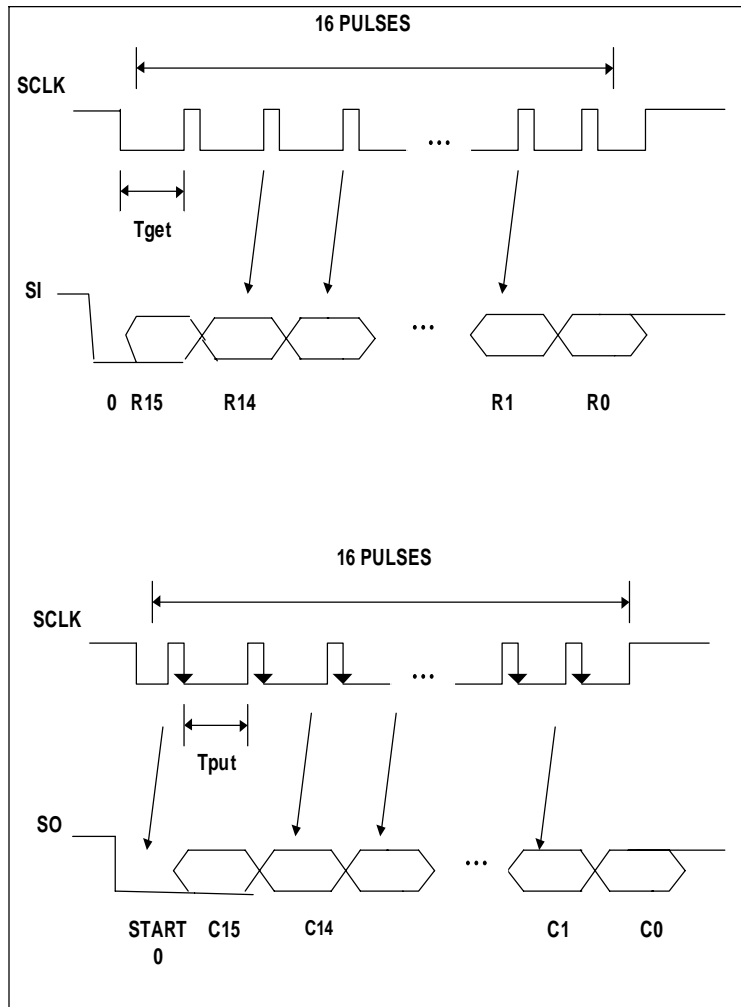
A 16-bit code can be accessed by HOST two times via setting PHILO pin. For writing operation, the HOST write the low byte of the command into the register of DSP, when PHILO is pulled to low, DSP does nothing but latches data. Following, PHILO is set to high and high byte of command is written into register, then DSP will perform the command issued by HOST and send out the response. DSP will pull PACK\ to be low to indicate that response is ready. HOST must read the low byte and then high byte sequentially. PACK\ will reset to high when the high byte is read by HOST. All of the description given above will be illustrated by the following timing chart.





**Synchronous Serial :**

The protocol of synchronous serial interface will be interpreted by the timing chart given below .



Note: SCLK : synchronous clock sent by HOST,  
SI : HOST serial in,  
SO : HOST serial out,  
 $T_{get}$  and  $T_{put}$  should be larger than 200  $\mu$ S

**Decision Rules between MCU mode and DSP Mode:**

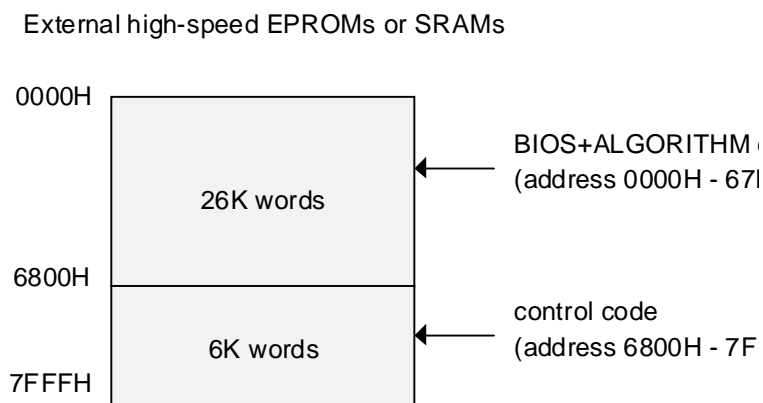
1. The contents located on 0xE000 and 0xE001 are 0x1234 and 0x5678, It will be considered as external DSP mode. This configuration is used for developing customer's control codes on MXIC's MX93032-M1 IC with standard code.
2. The contents located on 0x680E and 0x680F are 0x5678 and 0xA988, it will be considered as MCU mode. This is the only one configuration for MCU mode under developing and mass production.
3. Either the content located on 0x680E and not 0x5678 or that located on 0x680F are not 0xA988, it will be considered as DSP mode. The configuration is used for mass production for DSP mode adopted.
4. The decision priority depends on the items order. That is, the item labeled with smaller number has higher priority.

**CONTROL CODE CONFIGURATION (DSP MODE)**

There are two configurations for customer development or production.

**Configuration 1 :**

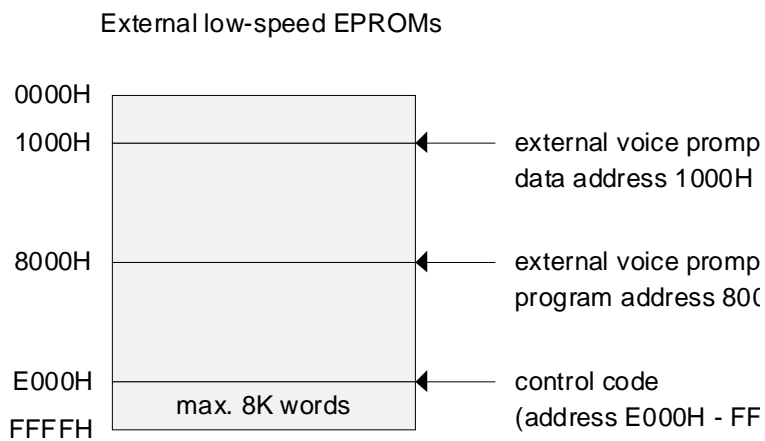
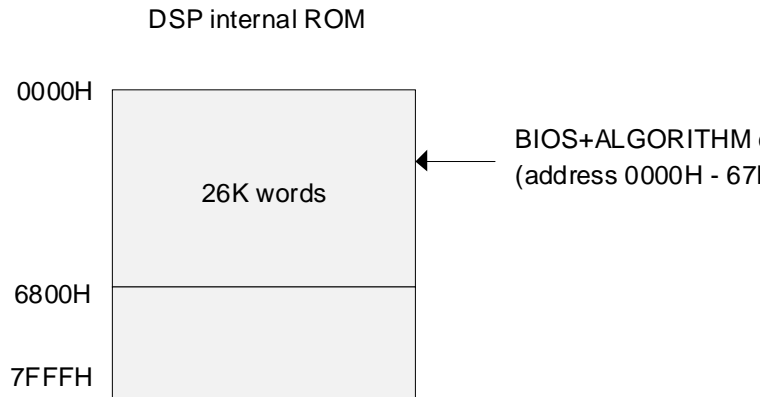
In this configuration, the BIOS+ALGORITHM and control codes are all running in the external high-speed EPROMs or SRAMs. The MX93032-M1 pin 97 (EROM) must be set to high.



This configuration is mainly used in the development stage and the access time of high-speed EPROMs or SRAMs must be less than 30ns.

**Configuration 2 :**

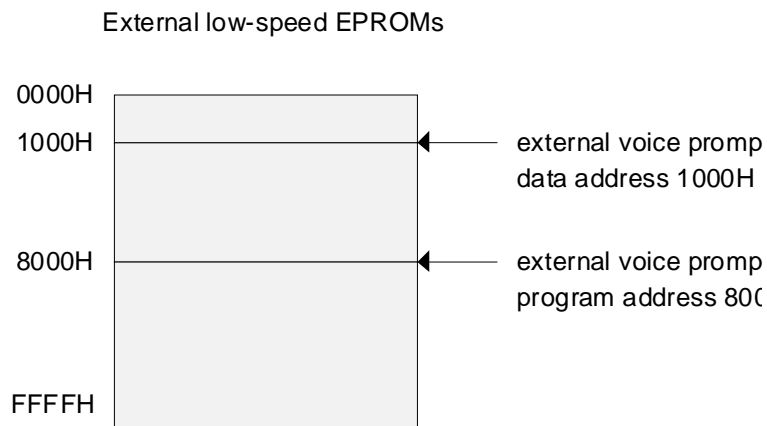
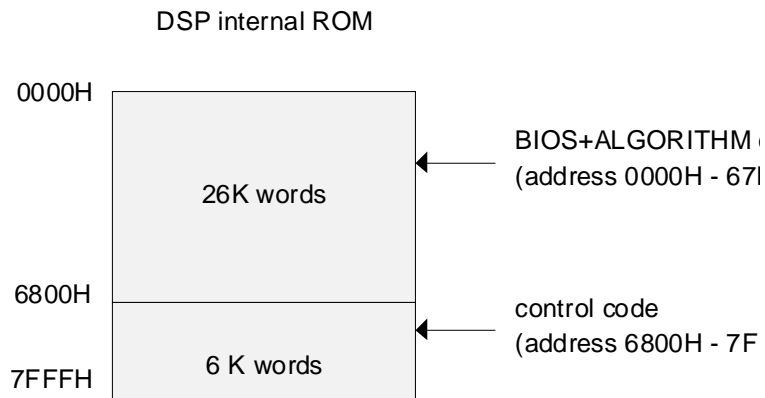
In this configuration, the BIOS+ALGORITHM code is running in the MX93032-M1 internal ROM and the control code running in the external low-speed EPROMs. The MX93032-M1 pin 97 (EROM) must be set to low.



This configuration can be used at the development stage or for production. The control code must start with two words 1234H and 5678H.

**Configuration 3 :**

In this configuration, the BIOS+ALGORITHM and control codes are all running in the MX93032-M1 internal ROM. The MX93032-M1 pin 97 (EROM) must be set to low.



This configuration is used for production.

To set the voice prompt configuration, see the MX93032-M1 Mode 7.

**MX93032-M1**

The MX93032-M1 provides DSP embedded functional modules, including speech compression/decompression, telephone line signal processing, ROM voice prompt, Macronix MX29F16A Flash management, time keeping and full duplex speaker phone.

In MCU mode, the MX93032-M1 service is requested via a command call with 16 bits parameter sent by Micro controller, and answered 16 bits sent by DSP. In DSP mode, user's system control program can be masked-in the MX93032.

The MX93032-M1 supports 16 modes of service and responses useful to operational information about timing, FLASH utilization, detected DTMF, etc.

**Hereafter, HOST means "external micro-controller" for MCU mode, and it means "system control program" for DSP mode.**

**FLASH MANAGEMENT**

The MX93032-M1 uses MXIC's MX29F16A FLASH as storage device for message and/or voice prompt data. Up to two MX29F16A can be supported by the MX93032-M1 service command.

The MX29F16A is a 16M-bit FLASH memory organized as 1M word x 16. It includes 16 sectors (each sector is 64K words) and some of these sixteen sectors are bad. To manage the MX29F16A, the MX93032-M1 will detect and then give up bad sectors and only utilize the good sectors. For the purpose of storage efficiency, each good sector is further divided into 32 blocks (each block contains 2K words). Each block can store speech data of about 6.8 seconds for 4.8kbps compression algorithm and 2.55 seconds for 12.8kbps compression algorithm.

**MESSAGE RECORDING AND STORAGE**

The MX93032-M1 provides high quality speech compression techniques for recording and playback functions. It can permit over 15 minutes or 40 minutes of speech storage in each 16M bit of MXIC's MX29F16A FLASH device based on 12.8K or 4.8K compression rate, respectively.

The MX93032-M1 supports up to 127 variable length in-

coming and outgoing messages that are labeled as Msg ID 1 to 127. Although ICM could range from Msg 1 to 127, Msg IDs 127 to 121 are the only places for outgoing messages (OGM) storage so that the deletion and recovery of old OGM are easier to maintain. The Msg IDs of incoming messages are labeled chronologically and rearranged automatically after recording or deleting. ICM with Msg ID 1 is the oldest recorded incoming message. Each message can be recorded by either 4.8K or 12.8K compression rate depending on system programmer's assignment.

During recording, the MX93032-M1 also monitors the telephone line signal to detect the presence of DTMF, Call Progress Tone and Continuous Tone and responses to the control program. The HOST can stop recording, and delete the last n\*400ms/200ms from the memory using a Record command with assigned Tail Cut factor. The MX93032-M1 can generate a desired tone during recording voice message, which allows the application such as two-way recording. During recording, Record Pause function is provided.

The recording voice signal, received through the CODEC input, is transmitted back to the CODEC output by the MX93032-M1 service commands.

**OUTGOING MESSAGE (OGM) RECORD**

The MX93032-M1 provides flexible recording ways for outgoing message. The HOST can select to delete the old OGM before a new one is recorded or to abort the new OGM record without losing the old OGM.

**NEW/OLD MESSAGE**

For the control program to manipulate incoming messages more handily, the MX93032-M1 keeps an internal record of the ICM to identify the new/old status of messages.

Before playback, the status of ICM is considered as "new". All incoming messages that have been played could change their status to "old" by issuing Real Del bit on Delete command. After Real Del is issued, all new/old ICM Msg IDs will be rearranged. The functions, Play New or Delete All Old messages, are provided by the MX93032-M1 in the command set.

## MESSAGE PLAYBACK

Random access for playback of any new message or message with specified Msg ID is supported by the MX93032-M1. During playback, the MX93032-M1 also monitors the telephone line signals, and Play Pause function is provided. The MX93032-M1 is able to fast-forward the playback of a recorded message, and the playing speed will be up to 33 %.

## MESSAGE DELETION

The MX93032-M1 provides two ways to delete recorded ICMs. One is the two-step method - the combination of Del Mark and Real Del on Play and Delete command, and the other is the immediate method - Delete command. The two-step method enables the control program to handle the deletion more flexibly, while the immediate method deletes a specified message directly.

## VOICE PROMPT

For high-quality voice prompt playback, the MX93032-M1 utilizes the built-in speech decoder and external EPROM/ROM or FLASH to store the vocabulary. The MX93032-M1 also supports up to a 32K-word external EPROM/ROM that allows minimum 1.8 minutes of voice prompt storage. If voice prompt data is stored in FLASH, space of maximum 60K words (minimum 3.4 minutes) is permitted. The external (upper part) voice prompt is also allowed. But if FLASH voice prompt is enabled, the external voice prompt will be disabled. The vocabulary supports up to 255 phrases. Given a phrase number in the Voice Prompt command from the HOST, the MX93032-M1 outputs the corresponding utterances to the Codec interface. The starting address of external voice prompt data could be located at data address 1000H, or program address 8000H.

During playback of the voice prompt, the MX93032-M1 monitors the telephone line signals.

## PERSONAL MAIL

The MX93032-M1 supports up to 8 mailboxes that may be used for personal mail application. Each mailbox has its own ICM/MEMO Msg ID. The total Msg.ID is 127 for all of mailbox. Before recording, playing or deleting mes-

sages, the HOST should set the corresponding mailbox number first with Personal Mail command.

## TIME KEEPING

There is a timer clock in the MX93032-M1. The HOST can issue Set Time and Get Time commands to get time services. The clock also supports system to stamp time to the recorded incoming message.

## DTMF DETECTION

For remote control operation of the answering machine, the MX93032-M1 monitors the incoming telephone line signal to detect the presence of DTMF signal.

## TONE GENERATION

Up to double tones can be generated by the MX93032-M1. The levels and frequencies are programmable by the HOST.

## CALL PROGRESS TONE DETECTION

The MX93032-M1 supports the detection of call progress tone within the band of 300 - 640 Hz for the incoming telephone line signal. The tone detector is implemented with a bandpass filter. Indication of the presence of call progress tone is responded to the control program, and could be used to terminate recording.

## NEAR END ECHO CANCELLATION

The MX93032-M1 implements a near end echo canceler in software to enhance the detection of DTMF and Call Progress Tone during Playback and Voice Prompt mode.

## FSK Decoder

All of the services, Calling Number Delivery (CND), Calling Name Delivery (CNAM) and Calling Identity Delivery on Call Waiting (CIDCW), exploit the Frequency Shift Keying (FSK) method to transmit the desired data from the SPCS to CPE. Depending on the specific service,

data transmission may occur with the CPE in an on-hook state or in an off-hook state. The specifications for data transmission are given on SR-TSV-002474 which are classified into three layers, physical layer, data linker layer and message assemble layer. The MX93032-M1 only takes the responsibility to decode physical layer. For Data link layer (Check Sum) and Message Assemble layer (the meaningful data), system should take care. The performance test associated with physical layer based on SR-3004 will be given on Appendix A.

Note: CPE: Custom Premises Equipment  
SPCS: Stored Program Controlled Switching System

### **CAS Detection**

For Calling Identity Delivery on Call Waiting, the MX93032-M1 will monitor the presence of CPE alerting signal according to the specification given on Bellcore Special Report SR-TSV-002476.

### **VOX DETECTION (DIGITAL VOICE ACTIVATED RECORDING)**

The speech detection is implemented on the MX93032-M1 to indicate whether the recording signal level is above the VOX threshold. If the control program sets the VOX Record bit on Initial command, the recording of the speech would start when the voice activity is detected. The HOST has to set VOX threshold by issuing command Mode 13. Hexadecimal numbers for VOX threshold are listed in Table 6.

### **SPEAKERPHONE**

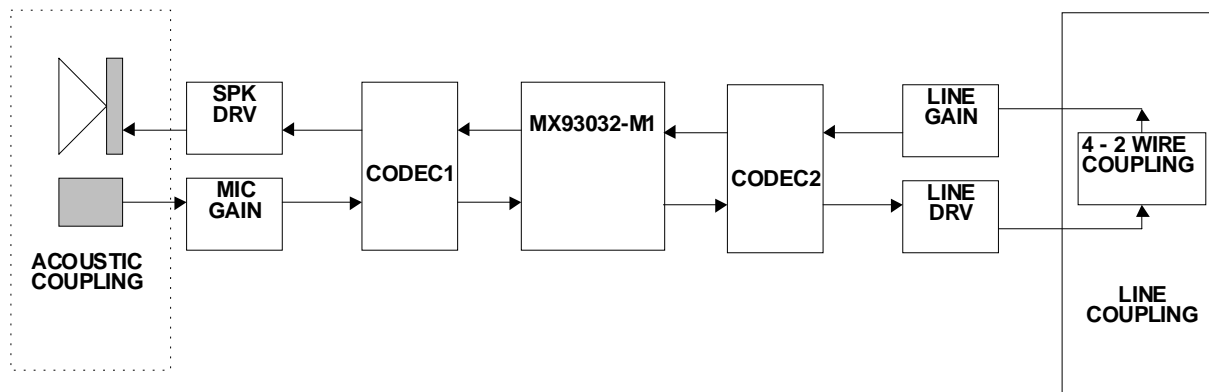
The speaker system provided by the MX93032-M1 contains two adaptive filters which are called as acoustical echo canceller (AEC) and line echo canceller (LEC). The purpose of AEC is used to cancel the acoustical echo between microphone and speaker which is caused by the sound reflection within space. A LEC cancels the echo resulted by the 4 to 2 wire telephone line connection and impedance mismatches between customer premises and the central office.

When speakerphone is activated, the algorithms associated with AEC and LEC not only reduce the echo heard

on the both ends, but also prevent feedback instability (singing) from occurring. Moreover, the MX93032-M1 provides automatic gain control (AGC) for microphone channel which helps the outgoing voice to keep at a desired level especially for a moving talker, and speaker volume control which allows the user to adjust the speaker volume for his need. In addition, DTMF/Tone generations and call-progress tone detection are also provided.

**DAA CONSIDERATION FOR SPEAKER PHONE**

It should be emphasized that DAA circuit plays an important role on improving the performance of speakerphone. This sub section gives some guides on how the DAA circuit may be "tuned". A diagram of DAA circuit must be given for simplifying the flowing interpretation.



Referring to figure given above, we have 4 analog amplifiers which are

1. Microphone Pre-Gain (MIC GAIN)
2. Line Out Gain (LINE DRV)
3. Line in Pre-Gain (LINE GAIN)
4. Speaker amplifier (SPK DRV)

Obviously, the 4 amplifiers together with the Line Out volume, Speaker Volume, receive attenuator, transmit attenuator, two volume compensators, acoustical coupling function  $H_{ac}(f)$  and line coupling function  $H_{lin}(f)$  compose a closed loop system. The AEC and LEC will provide additional attenuation to the loop gain one they converge.

To have a better performance on speaker phone, we strongly recommend that analog gain of DAA must satisfy the following two criteria:

1. As the near end (Microphone side) speaks only, the level at Codec 2 should fulfill that, "codec 2 input must be less than Codec 2 output due to electrical echo". This meaning can be expressed as

$$LINE\_DRV * H_{lin}(f) * LINE\_GAIN < 1$$

2. As the far end (Line Side) speaks only, the level at Codec 1 should fulfill that, "Codec 1 output must be greater than Codec 1 input resulting from acoustical echo". This meaning can be expressed as

$$SPK\_DRV * H_{ac}(f) * MIC\_GAIN < 1$$

By considering the criterion 1, it should pay more attention that minimum echo should be reached by improving the telephone line interface ( that is, reducing  $H_{lin}(f)$ ) and not by reducing  $LINE\_GAIN$  due to that the MX93032-M1 must obtain the high enough signal to process. For the same reason for criterion 2, the low echo should be achieved by good acoustical and mechanical decoupling rather than by decreasing  $MIC\_GAIN$ .

The ultimate target for hardware design is to create one system which not only satisfies two criteria given above, but also provides a desirable volume to both speaker and telephone.



## QUERY AND REPORT

The MX93032-M1 is capable of reporting to the control program about the query of the current status, such as the number of recorded messages, the number of new messages, the number of old messages, the available recording time of FLASH, the time of recorded message, and the FLASH good/bad check.

## POWER DOWN

With the power of backup batteries, the MX93032-M1 is capable of keeping messages and data stored in the FLASH during power failure. In power down mode, the MX93032-M1 will run at a lower clock rate to reduce power consumption.

## CRYSTAL CONNECTION

The MX93032-M1 allows two ways of crystal connections :

1. Normal approach: use both high crystal (32.256 MHz) and low crystal (32768 Hz). To enable this mode, the FLEN\ (54) pin must be connected to VDD. In this mode, the DSP high clock is generated from the high crystal directly, and the low crystal is mainly used in power down mode.
2. FLL (Frequency-Locked-Loop) approach: use only low crystal (32768 Hz). To enable this mode, the FLEN\ (54) pin and X2 (89) pin must be connected to ground and the X1 (88) pin to VDD. In this mode, the low crystal is to generate all the DSP system clocks and the high crystal can be saved to reduce the overall system cost.

## 7.0 OPERATIONAL DESCRIPTIONS

When the system is powered up, the RST\ pulse will make the MX93032-M1 to run programs from address 0. After check sum of internal ROM codes checked and some essential initialization of DSP are finished. In MCU mode, the MX93032-M1 is ready to poll the command issued by the HOST and indicate the HOST by setting PACK\ or SI\ to be low. The HOST should poll the indication sent by the MX93032-M1 and read out the pseudo response first. Following some necessary initialization given below must be set by HOST .

1. Set telephone data configuration
2. FLASH initialization depending on the FLASH good/bad check
3. Codec initialization if necessary
4. Setting Voice prompt configuration
5. Setting Silence threshold
6. Setting VOX level

## 8.0 MODES OF OPERATION

The MX93032-M1 supports 16 modes of operations, which are listed as follows :

Command Mode	Mode Name
0	Idle Mode
1	Record
2	Play
3	Message Status
4	DTMF Generator
5	Line Monitor
6	Delete Message
7	Set Current Time/Voice Prompt/Silence Threshold
8	Get Current Time
9	Initial and Test Memory
10	Get Record Message Information
11	Voice Prompt
12	Speaker Phone
13	Personal Mail
14	Telephone Data
15	CAS Detection

**IDLE :**

This command is allowable to be issued within any other operation which is processing. As soon as the MX93032-M1 receives Idle command, the original operation will be stopped automatically.

**RECORD**

When Record command is received from the HOST, the MX93032-M1 performs speech compression, stores the message into FLASH, and monitors telephone line. If the VOX Record bit had been set to 1 on Initial and Test Memory command, recording of the speech will start upon the voice activity detected, else the MX93032-M1 starts recording immediately after Record command is received.

Clearing Marked Record (bit 8-10) to 0, if it is recording ICM. When recording OGM, the HOST should set Marked Record (bit 8-10) to desired number other than 0 as OGM ID.

Setting bit 5 (Annc Del) to 1 on Initial and Test Memory command, the HOST can choose to delete the old OGM before a new one is recorded. If that bit is cleared to 0, the new OGM recording can be aborted without losing the old OGM by issuing Mark Fail (bit 11) on Record command.

During recording, the HOST can pause the recording by setting Pause (bit 7) to 1 and resume recording by clearing that bit. Line monitoring will continue during pause.

The HOST can stop recording by setting Stop (bit 6) to 1, and delete the last  $n \times 400\text{ms}/200\text{ms}$  from the memory by using a Record command with assigned Tail Cut factor (bit 0-5).

While system memory is full, MX93032-M1 will stop message recording, show the status in response word (bit 7, Mfull).

During Record mode, the HOST can instruct the MX93032-M1 to generate a tone by issuing the DTMF Generator command. Setting Stop bit on DTMF Generator command will stop tone generating.

The MX93032-M1 will terminate Record mode when Stop bit on Record command is set to 1.

Note: The MX93032-M1 supports up to 127 incoming and outgoing messages recording which are labeled as Msg ID 1-127. The Msg IDs of incoming message are labeled chronologically and rearranged automatically after recording. Although ICM could range from Msg ID 1 to 127, Msg IDs 127 to 121 are the only places for outgoing message storage. So, it is strongly suggested that the HOST should keep an upper bound of the number of ICM. (For example, upper bound of ICM = 127 - maximum OGM number that will be applied in user's specification.)

**PLAY**

When Play command is received from the HOST, the MX93032-M1 performs message playback and line monitoring. The Msg ID (bit 0-6) is set by the HOST to specify which message to play. To playback new message (non-played one), set Play New (bit 10) to 1, then the MX93032-M1 will search to play the non-played message with specified Msg ID. During playing, the HOST can pause playback by setting Pause (bit 8) to 1 and resume playing from the same point by clearing that bit. Line monitoring will continue during pause. To fast playback a recorded message, set Fast Play (bit 11) to 1 and the playing speed will be up to 33%.

To delete and stop the playing message, the HOST can set Del Mark (bit 7) on Play command. After being tagged a "Del Mark" the marked message is not really deleted until a Real Del (Delete Message mode bit 8) is set. The Real Del command not only deletes the messages that have been tagged a "Del Mark" but also changes the status of those ICMs that have been played to old. The ICMs that have not been played before are regarded as "new". After Real Del is issued, all new/old ICM Msg IDs will be rearranged.

For the HOST, it is used to issue the Real Del after the user terminates the manner of playback.

The response word contains the information of tones detected from the line, playing time spent (bit 8-15) and End of play flag (bit 6), which indicates the end of the current message. When the Play End bit is 1, the MX93032-M1 stops playback.

The MX93032-M1 will terminate Play mode when Stop bit or Del Mark bit is set to 1 on Play command.

### MESSAGE STATUS

In this mode, there are 6 request codes for the HOST to get message information;

1. Get the number of personal recorded messages (request code = 0)
2. Get the number of personal new messages (request code = 1)
3. Get the number of personal old messages (request code = 2)
4. Get the available recording time with unit of second (request code = 3)
5. Get the number of total recorded messages (request code = 8)

The result will be reported in the response word.

### DTMF GENERATOR

In this mode, up to double tones can be generated by the MX93032-M1. The tone levels and frequencies are programmable and controlled by the HOST. The DTMF Generator command defines the output gains in bit 0-3 for the 1st tone, and bit 4-7 for the 2nd tone. Each tone can be programmed from 3 dB (gain code 0000) to -25 dB (gain code 1110) with 2 dB resolution. For DSP mode, the frequencies are defined in the variables BUF1 and BUF2. For MCU mode, three steps of commands should be issued, first, DTMF sent and signal Gain should be set, following the corresponding two frequencies should be specified. The frequency code and its frequency represented is according to the following relationship :

frequency code = desired frequency (Hz) \* 8.19

$f(n) = 32767 * \cos(2 * \pi * f\_code * n / (8000 * 8.19))$

where f\_code is the input frequency code, n is sequence index and 8000 is the sampling rate

For single tone, Codec outputs the signal s(n) :

$$s(n) = \text{gain} * f(n)$$

For dual tone, Codec outputs the signal s(n) :

$$s(n) = 0.5 * (\text{gain1} * f1(n) + \text{gain2} * f2(n))$$

To generate a single tone, the gain code of the other tone should be set to 1111. The MX93032-M1 starts tone generation only after receiving the start command.

The MX93032-M1 will terminate DTMF Generator when Stop bit is set to 1 on DTMF Generator command.

**LINE MONITOR**

In this mode the MX93032-M1 monitors the telephone line signal for the detection of DTMF, Call Progress Tone, VOX, and/or Continuous Tone. The MX93032-M1 keeps monitoring the line until Stop bit is set to 1.

The MX93032-M1 returns to the HOST a response word that reflects the flag of Call Progress Tone, VOX, Continuous Tone and the index of DTMF signal that has been detected :

INDEX		DTMF CODE	INDEX		DTMF CODE
CAS	DTMF		CAS	DTMF	
0	0000	No Tone	0	1001	9
0	0001	1	0	1010	*
0	0010	2	0	1011	0
0	0011	3	0	1100	#
0	0100	4	0	1101	A
0	0101	5	0	1110	B
0	0110	6	0	1111	C
0	0111	7	1	1111	D
0	1000	8	1	0000	CASTONE

The MX93032-M1 will terminate Line Monitor mode when Stop bit is set to 1 on Line Monitor command.

**DELETE MESSAGE**

This mode provides an immediate way to delete messages as follows;

1. To delete a specified message, set its Msg ID in bit 0-6 and clear bit 11, 10 and 7 to zero.
2. To delete a specified new ICM, set Del New Msg (bit 10) to 1, give the New Msg ID in bit 0-6, clear bit 11, 7 to zero.
3. To delete all old ICM, set Del All Old (bit 7) to 1 and clear bit 11, 10 to zero. (Bit 0-6 is ignored)
4. To erase the "Del Mark" from a specified ICM ("Del Mark" is a tag set on Play command for deletion), set Del Mark Clr (bit 11) to 1, give the Msg ID in bit 0-6 and clear bit 7 to zero.

**SET CURRENT TIME/VOICE PROMPT/SILENCE THRESHOLD**

In this mode the HOST can set the current time to the timer clock of the MX93032-M1. The time information includes second, minute, hour and week. The timer in the MX93032-M1 will be used to stamp time and data to the recorded message.

1. To set second, put the data in bit 0-7 and use request code (bit 8-11) 0000.
2. To set minute, put the data in bit 0-7 and use request code (bit 8-11) 0001.
3. To set hour, put the data in bit 0-7 and use request code (bit 8-11) 0010.
4. To set week, put the data in bit 0-7 and use request code (bit 8-11) 0011.
5. To set voice prompt configuration, put the selection data in bit 0-1 and use request code (bit 8-11) 0110.
6. To set silence threshold level, put the data in bit 0-3 and use request code (bit 8-11) 0111.
7. To set output port OPT0-7, put data in bit0-7(Request Code=0100)
8. To set codec MX93002 registers in bit10-8 and put data in bit0-7(Request Code=1xxx)
9. To write link index and data in bit0-7(Request Code=0101), refer to mode10 to specify Msg. ID.

## GET CURRENTTIME

In this mode the HOST can get the current time from the timer clock of the MX93032-M1. The time information includes second, minute, hour and week.

1. To get second, use request code (bit 8-11) 0000.
2. To get minute, use request code (bit 8-11) 0001.
3. To get hour, use request code (bit 8-11) 0010.
4. To get week, use request code (bit 8-11) 0011.

The requested data will be reported in the response word.

## INITIAL AND TEST MEMORY

In this mode there are 9 functions for the control program to define;

### 1. FLASH Re-Initialization

The HOST can re-initialize the FLASH by setting FLASH Re-Init (bit 0) to 1. After receiving this command, the MX93032-M1 will erase all the messages and data stored in the FLASH and then re-format the FLASH to establish the management structure. The response word reported to the control program contains the information of FLASH bit size (bit 0-3), FLASH address size (bit 4-7), FLASH good/bad status (bit 8) and FLASH good rate (bit 9-15). After re-initialization, all messages and data in the FLASH are lost.

### 2. FLASH Initialization or Error Correction

The HOST can perform FLASH first-time initialization or error correction by setting FLASH Init (bit 1) to 1. First-time initialization is to format a new FLASH and then establish the management structure, while error correction is to correct the problems that may result from power loss during operation. This function is normally used after first power-up of the system. The response word reported to the control program contains the information of FLASH bit size (bit 0-3), FLASH address size (bit 4-7), FLASH good/bad status (bit 8) and FLASH good rate (bit 9-15).

### 3. FLASH Garbage Collection

Due to the characteristics of MX29F16A FLASH and the management structure, when messages are deleted, the data regions occupied by them will not be released to be available immediately. To free out such regions, the MX93032-M1 provides a two-step mechanism. First is to check the FLASH garbage collection recommendation status by Message Status command and, if garbage collection is recommended, then to issue bit 2 (Garbage Collect) on this command to perform the corresponding garbage collection actions.

### 4. Line On/Off

The HOST should set Line On/Off (bit 3) to 1 if the system is logged in from telephone line, and clear that bit to 0 if in local keypad operation. Failing to set this bit properly will degrade the performance of echo cancellation.

### 5. VOX On/Off Record

If the HOST turns on the VOX Record function (by setting bit 4 to 1) on this command, speech recording will start upon the voice activity detected, else recording will start immediately after Record command is issued.

#### 6. OGM Deletion

By setting bit 5 (Annc Del) to 1 on this command, the HOST can choose to delete the old OGM before recording a new one. If that bit is cleared to 0, the old OGM is not deleted until the new OGM recording is completed. With this option, the new OGM recording can be aborted without losing the old OGM by issuing Mark Fail (bit 11) on Record command.

#### 7. New/Old Select

By setting New/Old Select (bit 6), the HOST can decide which condition a new message will be changed to an old one. There are two options. When this bit is set to 1, the new message that has been played (even not played end) is regarded as an old message. If this bit is cleared to 0, the new message that has been played end is regarded as an old one.

#### 8. Tail Cut Unit

By setting Tail Cut Unit (bit 7) to 1, the HOST can choose the tail cut unit length to be 200 ms. If this bit is cleared to 0, the unit length is 400 ms.

#### 9. Old Mark Set

By setting Old Mark Set (bit 8) to 1, the HOST can disable that new messages change to be old ones even though they have been played (end). If this bit is cleared to 0, new to old change will be enabled and the change criterion is according to the setting of New/Old Select bit (bit 6).

### GET RECORD MESSAGE INFORMATION

In this mode, there are 10 request codes for the control program to get the related information about a specified message.

Given the Msg ID (bit 0-6) and New Msg flag (bit 7), the MX93032-M1 can provide message recording time, recording length, attribute and "Del Mark" status according to the Request Code (bit 8-11) as follows :

1. Get the recording time - second. (Request Code = 0000)
2. Get the recording time - minute. (Request Code = 0001)
3. Get the recording time - hour. (Request Code = 0010)
4. Get the recording time - week. (Request Code = 0011)
5. Get the recording length (sec.). (Request Code = 0100)
6. Get the timer status. (Request Code = 0101). The response word is 1 if the MX93032-M1 timer has been set before with Set Current Time command. Otherwise, the response word is 0.
7. Get the message ICM/MEMO attribute. (Request code = 0110). To show the specified message is of MEMO type (response word = 1) or ICM type (response word = 0).
8. Get the Del Mark status (Request Code = 0111). To check if the specified message has been tagged a Del Mark (response word = 1) or not (response word = 0).
9. Get the message new/old status. (Request Code = 1000). To check if the specified message has been tagged an Old Mark (response word = 1) or not (response word = 0). A new message will be tagged an Old Mark when it has been played/played end and changes to old messages as Real Del command is issued.
10. Get the OGM existing status (Request Code = 1111). To check if the specified OGM exists (response word = 1) or not (response word = 0). Msg ID could only range from 127 to 121.
11. To get link index data (Request Code=1001), refer to mode7
12. To set the Msg. ID for reading/writing Link-index-data. (Request Code=1010)

All the results are reported in the response word.

## VOICE PROMPT

In this mode the MX93032-M1 plays back a speech segment that the corresponding phrase number is given in command bit 0-7. Up to 255 speech segments, previously stored in voice prompt ROM or Flash memory, are provided by the MX93032-M1 and pronounced using the speech decompression algorithm. The HOST can pause the playback by setting Pause (bit 8) to 1 or resume from pause by clearing Pause bit to 0. Fast forward playback of voice prompt is also provided by setting Fast Play (bit 11) to 1. During Voice Prompt mode, the MX93032-M1 keeps monitoring the line signal and sends the results in response word to the control program.

The MX93032-M1 will terminate Voice Prompt mode when Stop bit is set to 1 on Voice Prompt command.

## SPEAKERPHONE

Regarding to speakerphone related function is comprised of three types which are parameter setting, DTMF/Tone generation and free run. Two Codecs are necessary for speakerphone operation. One Codec labeled as Codec 1 is used for microphone and speaker, and the other labeled as Codec 2 is used for telephone line input and output. In this mode, there are seven functions whose detail will be given as follows:

1. Free run which performs both filtering operation and updating operation on LEC and AEC one time for each Codec interrupt triggered (Request Code = 0x0000)
2. Setting the gain for DTMF/Tone generation (Request Code = 0x0001)
3. Generating DTMF/Tone according to pre-defined table (Request Code = 0x0010)
4. Setting both the echo return-loss on speaker side (ERL\_AEC) and echo return loss on line side (ERL\_LEC) (Request Code = 0x0100)
5. The T/R and R/T ratio are used to decide which side is speaking by comparing the transmitting and receiving energy.(Request Code=0x0110)
6. Set the volume gains to control the speaker out and line out signal levels(Request Code=0x1000)
7. The loop attenuation is to control the total receive and transmit attenuators (Request Code = 0x1001)
8. setting the tone table which require 3 steps : Index set, Freq\_Lo set and Freq\_Hi set.

## PERSONAL MAIL

Five functions are provided in this command as follows :

1. Set the personal mailbox number

In this mode the HOST can select one out of eight mailboxes (setting in bit 0-2) for recording messages, playing recorded messages and retrieving message time. Each mailbox manages its own messages and Msg IDs. Make sure to set the desired mailbox before issuing the message related commands such as Record, Play, Delete Message and Get Record Message Information.

2. Set the recording message attribute

Before recording a message, the HOST can set the attribute of that message to be MEMO or ICM. Clear bit 3 to zero for recording a message as an ICM or set bit 3 to one for recording a message as a MEMO.

3. Set the digital loop back control

The HOST can enable or disable the DSP digital loop back path in Record or Line Monitor operation. By setting Loop Back bit to 1, the loop back path is disabled and the signal received from the codec input will not be transmitted back



to the codec output. When this bit is cleared to 0, loop back is enabled and the received signal will be transmitted back to the codec output. The default status is the latter.

#### 4. Set the output level

The HOST can set the level difference between the output (playback) speech and the corresponding input (recorded-in) speech. By setting the Output Level bit to 1, the output speech level is equal to the input speech level, and clearing this bit to 0, the output speech level is about 5dB less than the input speech level. The default status is the latter.

#### 5. Set the codec data format

The HOST can select one of two Codec data formats by setting the Codec Format bit. When Codec Format is cleared to 0, 8-bit m-law data format is selected, and set to 1, 16-bit linear data format is enabled. At present, MXIC's Codec family provide both m-law and linear formats, and the latter is suggested because of the better speech quality. The default format is 8-bit m-law.

#### 6. Set Voice Activity

In this mode the HOST can set the VOX threshold in bit 0-11. The hexadecimal numbers for VOX level are listed in Table 6.

#### 7. Set Telephone Data Configuration

This command should be issued before Aflash initialization.

### CAS Detection

The MX93032-M1 provides 2 output modes for CAS detection, which are continuous response and one time response.

For continuous response, the MX93032-M1 will respond CAS detected after  $45 \pm 5$ ms since CAS occurs in CPE.

Regarding to response one time, the information of CAS detected will be responded after both the CAS is disappeared on the line and the duration for CAS appearance between 45ms to 95 ms.

### FSK Detection

The output of FSK detector can be divided into two mode which are Raw data mode and Cooked data buffer. The buffer length for Raw data mode and Cooked Data mode are 8 bytes and 60 bytes long, respectively. For Raw data mode, the data ready bit will be set on one bytes data ,in which start bit and stop bit are not included, received correctly. If the host's speed can not catch up the FSK baudrate, buffer overflow will occur. Therefore, when the serial interface is adopted, we do not recommend to select Raw data mode to receive FSK data. Cooked data mode concerned, the data ready bit will be set after all of the data, in which checksum is included, received. During the operation for Cooked data mode, the second data byte following the mark data will be considered as the total data length, the other data will not be recognized by MX93032-M1.

### LINKAGE BETWEEN CID DATA AND VOICE MESSAGE

1. Set the link index which can be also considered as CID data number (Request code = 0x7500 | link index)
2. Get the link index (Request code = 0xA900 | msg\_no)
3. write the link index specified by 0x7500 into message (Request code = 0xAA00 | msg\_no)

**TELEPHONE DATA**

In this mode the HOST can write/read telephone or user-assigned data to/from the FLASH with the sequence of telephone data command as follows :

**1. Write Telephone Data**

The first command defines the telephone record ID (bit 0-7) and set bit 11, 10 to 0, 0.

Following command specifies the one-byte data to be stored under that ID.

While the current record ID is full, the MX93032-M1 BIOS will show the status in response word (bit 0, Full).

The MX93032-M1 BIOS will terminate Telephone Data mode when Stop (bit 9) is set to 1 on Telephone Data command.

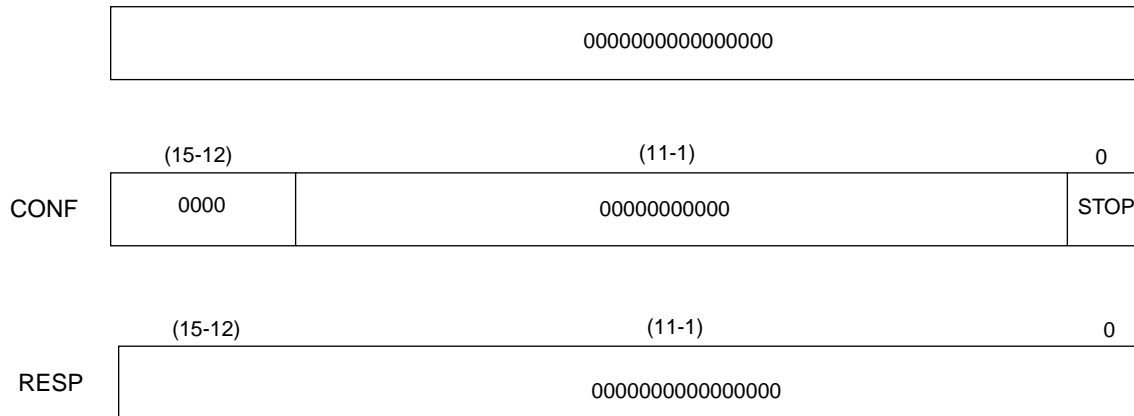
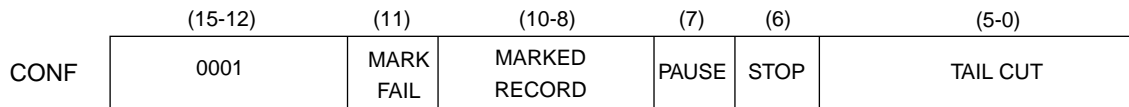
**2. Read Telephone Data**

The first command defines the telephone record ID (bit 0-7) and set bit 11, 10 to 0, 1. The telephone data will be put on the response word. Following command puts the one-byte data on the response word to the control program.

The response word contains the byte-data (bit 0-7) and Read End (bit 8). When the Read End bit is 1, the byte-data is invalid.

The MX93032-M1 BIOS will terminate Telephone Data mode when Stop (bit 9) is set to 1 on Telephone Data command.

If host want to write/read more than 2 Tel. Record IDs at the same time, Before writing/reading an new Tel. ID, host should set STOP bit to 1 for the current Tel. ID.

**9.0 COMMAND FORMAT AND PROTOCOL**
**Mode 0. IDLE MODE**

**Mode 1. RECORD MODE**


TAIL CUT : 0 ~ 63 units. One unit length can be 400ms or 200ms (set by Mode 9).

STOP = 1 : stop recording.

PAUSE = 0 : continue recording.

= 1 : pause recording.

**MARKED RECORD**

= 000 : record a general message.

= 001 : record OGM1 and store it in message 127.

= 010 : record OGM2 and store it in message 126.

= 011 : record OGM3 and store it in message 125.

= 100 : record OGM4 and store it in message 124.

= 101 : record OGM5 and store it in message 123.

= 110 : record OGM6 and store it in message 122.

= 111 : record OGM7 and store it in message 121.

MARK FAIL =1 : give up and stop recording of the new OGMx and recover the original OGMx. If Mode 9 ANNC DEL=1, this bit will be ignored.

When MARKED RECORD = 000, do not set this bit.

	(15-9)	(8)	(7)	(6)	(5)	(4)	(3-0)
RESP	RECORD TIME (sec)	CAS	MFULL	VOX	TONE	CONT TONE	DTMF

DTMF : the detected DTMF value during recording.

CONT TONE = 1 : continuous tone is found.

TONE = 1 : call progress tone is found.

VOX = 1 : the input speech power level is under the VOX LEVEL (set by Mode 12).

MFULL = 1 : ARAM is full and no further recording is possible. In this case, one more STOP command is needed to completely stop the recording action.

RECORD TIME : recording time until now, 0 ~ 127 sec.

CAS : CAS detection for CIDCW (detail see mode 15)

If CAS tone and DTMF D are existed at the same system, the Host should Check if bit8(CAS) == 1 and bit3-0(DTMF) = 0000, and then that's a real CAS Tone. Refer to the following table.

**DTMF Reference Table:**

INDEX		DTMF CODE	INDEX		DTMF CODE
CAS	DTMF		CAS	DTMF	
0	0000	No Tone	0	1001	9
0	0001	1	0	1010	*
0	0010	2	0	1011	0
0	0011	3	0	1100	#
0	0100	4	0	1101	A
0	0101	5	0	1110	B
0	0110	6	0	1111	C
0	0111	7	1	1111	D
0	1000	8	1	0000	CAS TONE

**Mode 2. PLAY MODE**

	(15-12)	(11)	(10)	(9)	(8)	(7)	(6-0)
CONF	0010	FAST PLAY	PLAY NEW	STOP	PAUSE	STOP	MSG ID

- MSG ID = 1 ~ 127 can be assigned for playing.
- DEL MARK = 1 : stop playing and mark the current message with a "Del Mark". This message will not be deleted until the Mode 6 REAL DEL command is done.
- PAUSE = 0 : continue playing.  
= 1 : pause playing.
- STOP = 1 : stop playing.
- PLAY NEW = 0 : the MSG ID is related to total messages.  
= 1 : the MSG ID is related to new messages.
- FAST PLAY = 0 : normal play speed.  
= 1 : play speed will be up to 33%.

	(15-9)	(8)	(7)	(6)	(5)	(4)	(3-0)
RESP	PLAY TIME (sec)	CAS	0	PLAY END	TONE	0	DTMF

- DTMF and TONE : see RECORD MODE.
- PLAY END = 1 : the current message is played end.
- PLAY TIME : playing time until now, 0 ~ 239 sec.
- CAS : CAS detection for CIDCW (detail see mode 15 and mode1)

**Mode 3. MESSAGE STATUS MODE**

	(15-12)	(11-4)	(3-0)
CONF	0011	0000000000	REQUEST CODE

	(15-0)
RESP	DATA

**REQUEST CODE:**

REQUEST CODE	STATUS	DATA
0000	number of personal total messages	0 ~ 127
0001	number of personal new messages	0 ~ 127
0010	number of personal old messages	0 ~ 127
0011	available time	0 ~ 65535
0101	garbage collection recommendation	0 or 1
1000	number of total messages	0 ~ 127

**Mode 4. DTMF GENERATOR MODE**
**For DSP mode:**

	(15-12)	(11)	(10)	9	8	(7-4)	(3-0)
CONF	0100	START	STOP	TO	0	GAIN2	GAIN1

GAIN1, GAIN2 : 0 ~ 14 denote 3dB ~ -25dB (2dB/one step), 15 denotes no related signal.

TO =0 OUTPUT CHANNEL CODEC 0.

=1 OUTPUT CHANNEL CODEC1.

STOP = 1 : stop the DTMF generation.

START = 1 : start the DTMF generation. When this bit is set, GAIN1 and GAIN2 must be also set and Frequency 1 is put in BUF1 (data address 7) and Frequency 2 in BUF2 (data address 8).

16-bit Frequency = target frequency \* 8.192

**Note :**

For single-tone signals (GAIN1 = 15 or GAIN2 = 15), the output level is according to GAIN1 or GAIN2. But for dual-tone signals, the output level is equal to half of sum of two individual signals. Doing this can avoid saturation of output signals.

	(15-0)
RESP	0000000000000000

**For MCU mode:**

	(15-12)	(11)	(10)	9	8	(7-4)	(3-0)
CONF	0100	START	STOP	TO	0	GAIN2	GAIN1

GAIN1, GAIN2 : 0 ~ 14 denote 3dB ~ -25dB (2dB/one step), 15 denotes no related signal.

TO =0 OUTPUT CHANNEL CODEC 0.

=1 OUTPUT CHANNEL CODEC1.

STOP = 1 : stop the DTMF generation.

START = 1 : start the DTMF generation. When this bit is set, GAIN1 and GAIN2 must be also set and Frequency 1 and Frequency 2 is put in the following command.

16-bit Frequency = target frequency \* 8.192

**Note :**

For single-tone signals (GAIN1=15 or GAIN2=15), the output level is according to GAIN1 or GAIN2. But for dual-tone signals, the output level is equal to half of sum of two individual signals. Doing this can avoid saturation of output signals.

(15-0)

RESP 

0000000000000000
------------------

(15-0)

CONF 

Frequency 1
-------------

(15-0)

RESP 

0000000000000000
------------------

(15-0)

CONF 

Frequency 2
-------------

(15-0)

RESP 

0000000000000000
------------------

**Mode 5. LINE MONITOR MODE**

	(15-12)	(11-8)	(7)	(6-1)	(0)
CONF	0101	0000	PAUSE	000000	STOP

STOP = 0 : start and continue the line monitor mode.  
 = 1 : stop the line monitor mode.

	(15-9)	(8)	(7)	(6)	(5)	(4)	(3-0)
RESP	0000000	CAS	0	VOX	TONE	CONT TONE	DTMF

DTMF, CONT TONE, TONE, PAUSE, VOX and CAS: see RECORD MODE.

**Mode 6. DELETE MESSAGE MODE**

	(15-12)	(11)	(10)	(9)	(8)	(7)	(6-0)
CONF	0110	DEL MARK CLR	DEL NEW MSG	0	REAL DEL	DEL ALL OLD	MSG ID

MSG ID : 1 ~ 127.

DEL ALL OLD = 1 : delete all old messages. (in this case, the MSG ID field is ignored)

REAL DEL = 1 : delete the messages that have "Del Mark" and change the new messages that have been played to be old messages.  
(in this case, the MSG ID field is ignored)

DEL NEW MSG = 0 : the MSG ID is related to total messages.

= 1 : the MSG ID is related to new messages.

DEL MARK CLR= 1 : clear the message "Del Mark" (set by Mode 2 DEL MARK=1).

	(15-12)	(11-0)
RESP	0110	000000000000

**Mode 7. SET CURRENT TIME/VOICE PROMPT/SILENCE THRESHOLD MODE**

	(15-12)	(11-8)	(7-0)
CONF	0110	REQUEST CODE	DATA

	(15-12)	(11-0)
RESP	0111	000000000000

**REQUEST CODE:**

TIME	REQUEST CODE	DATA
SECOND	0000	0 ~ 59
MINUTE	0001	0 ~ 59
HOUR	0010	0 ~ 23
WEEK	0011	0 ~ 6
SET OUTPUT PORT	0100	00 ~ FF H
SET LINK INDEX	0101	00 ~ FF H
VOICE PROMPT CONFIG	0110	0 ~ 7
SILENCE THRESHOLD	0111	0 ~ F H
SEND MX93002 CMD 93993000(A) CMD	1xxx	00 ~ FF H



**Request Code = 0110** (set voice prompt configuration),

	(15-12)	(11-8)	(7-3)	(2)	(1)	(0)
CONF	0110	0110	00000	FVOP SELECT	EVOP SELECT	IVOP SELECT

- IVOP SELECT (internal voice prompt setting),
  - = 0 : internal voice prompt not selected.
  - = 1 : internal voice prompt selected and test.
- EVOP SELECT (external voice prompt setting),
  - = 0 : external voice prompt not selected.
  - = 1 : external voice prompt selected and test.
- FVOP SELECT (FLASH voice prompt setting),
  - = 0 : FLASH voice prompt not selected.
  - = 1 : FLASH voice prompt selected and test.

	(15-5)	(4)	(3)	(2)	(1)	(0)
RESP	00000000000	FVOP STATUS	EVOP UNIT	EVOP ADDR	EVOP STATUS	IVOP STATUS

- IVOP STATUS (internal voice prompt test status),
  - = 0 : test fail.
  - = 1 : test ok.
- EVOP STATUS (external voice prompt test status),
  - = 0 : test fail.
  - = 1 : test ok.
- FVOP STATUS (FLASH voice prompt test status),
  - = 0 : test fail.
  - = 1 : test ok.
- (when EVOP STATUS = 1),
  - EVOP ADDR = 0 : external voice prompt at data space 1000H.
  - = 1 : external voice prompt at program space 8000H.
- EVOP UNIT = 0 : external voice prompt in byte unit.
- = 1 : external voice prompt in word unit.

**Request Code = 0111** (set silence threshold level - used by the silence management algorithm),

	(15-12)	(11-8)	(7-4)	(3-0)
CONF	0111	0111	0000	SILENCE LEVEL

SILENCE LEVEL : 0 ~ 15, the larger the level is, the more speech will be recognized as silence.  
 0 denotes no silence management.  
 Please refer to Table 7 for Silence Threshold Level.

	(15-0)
RESP	0000000000000000

**Mode 8. GET CURRENT TIME MODE**

	(15-12)	(11-8)	(7-0)
CONF	1000	REQUEST CODE	00000000
	(15-0)		
RESP	DATA		

REQUEST CODE, 0000 ~ 0011 : see Mode 7.

**Mode 9. INITIAL AND TEST MEMORY MODE**

	(15-12)	(11-9)	(8)	(7)	(6)	(5)	(4)	(3)	(2)	(1)	(0)
CONF	1001	000	OLD MARK SET	TAIL CUT UNIT	NEW/ OLD SELECT	ANNC DEL	VOX ON/OFF RECORD	LINE ON/ OFF	GARBAGE COLLECT	FLASH INIT	FLASH RE-INIT

FLASH RE-INIT = 1 : re-initialize the FLASH. The information of FLASH size and FLASH good rate will be responded in RESP. (Before this command is issued, Tel. Data Config., BIOS Mode 13 Sub-mode 3, should be set properly.)

FLASH INIT = 1 : do FLASH first-time initialization or error correction after power loss. The information of FLASH size and FLASH good rate will be responded in RESP. (Before this command is issued, Tel. Data Config., BIOS Mode 13 Sub-mode 3, should be set properly.)

GARBAGE COLLECT = 1 : perform FLASH garbage collection. After garbage collection, more deleted (but not released) data regions will be free out for further use.

LINE ON/OFF = 0 : not in remote line operation.  
= 1 : in remote line operation.

VOX ON/OFF RECORD = 0 : record the starting silence.  
= 1 : not record the starting silence which power level is under VOX LEVEL.

ANNC DEL = 0 : delete the old OGMx after the new OGMx is recorded.  
= 1 : delete the old OGMx before the new OGMx is recorded.

NEW/OLD SELECT (set the judgment point of message new/old attributes)  
= 0 : one message that had been played end is viewed as an old message.  
= 1 : one message that has been played (even not played end) is viewed as an old message.

TAIL CUT UNIT (set the tail cut unit)  
= 0 : the tail cut unit is 400 ms.  
= 1 : the tail cut unit is 200 ms.

OLD MARK SET = 0 : enable setting old mark.  
= 1 : disable setting old mark. New messages will not change to be old even they have been played (end). In this case, NEW/OLD SELECT will be ignored.

	(15-9)	(8)	(7-4)	(3-0)
RESP	FLASH GOOD RATE	FLASH G/B	FLASH ADDR SIZE	FLASH BIT SIZE

FLASH BIT SIZE : the detected FLASH data bit number in one address access.

FLASH ADDR SIZE : the detected FLASH address space.

FLASH G/B (a conclusion of FLASH initialization or error correction),

= 0 : FLASH is bad.

= 1 : FLASH is good.

FLASH GOOD RATE : 0 ~ 100 %. It is significant when FLASH G/B = 1.

CODE	FLASH ADDR SIZE	CODE	FLASH BIT SIZE
0000	0 M	0000	0
0101	1 M	1111	16
0110	2 M		

#### Mode 10. GET RECORD MESSAGE INFORMATION MODE

	(15-12)	(11-8)	(7)	(6-0)
CONF	1010	REQUEST CODE	NEW MSG	MSG ID

MSG ID : 1 ~ 127.

NEW MSG = 0 : the MSG ID is related to total messages.

= 1 : the MSG ID is related to new messages.

INFORMATION	REQUEST CODE	DATA
SECOND	0000	0000 ~ 003BH
MINUTE	0001	0000 ~ 003BH
HOUR	0010	0000 ~ 0017H
WEEK	0011	0000 ~ 0006H
RECORD LENGTH	0100	BIT 15 - BIT 0
TIMER STATUS	0101	0 = default timer ; 1 = timer has set
ICM/MEMO	0110	0 = ICM record ; 1 = MEMO record
DEL MARK STATUS	0111	0 = no del mark ; 1 = has del mark
MSG NEW/OLD STATUS	1000	0 = no old mark ; 1 = has old mark
GET LINK INDEX	1001	00~FFH
WRITE LINK INDEX (specified by 75XX)	1010	MSG ID=1~127
OGM EXISTING STATUS	1111	0 = OGM not exist ; 1 = OGM exist

	(15-0)
RESP	DATA

**Mode 11. VOICE PROMPT MODE**

	(15-12)	(11)	(10)	(9)	(8)	(7-0)
CONF	1011	FAST PLAY	0	STOP	PAUSE	PHRASE NUMBER

PHRASE NUMBER : 1 ~ 255.

PAUSE, STOP and FAST PLAY : see PLAY MODE.

	(15-9)	(8)	(7)	(6)	(5)	(4)	(3-0)
RESP	PLAY TIME(sec)	CAS	0	PLAY END	TONE	0	DTMF

DTMF, TONE, PLAY END CAS and Play TIME : see PLAY MODE.

**Mode 12. SPEAKERPHONE MODE**

	(15-12)	(11-8)	(7-0)
CONF	1100	REQUEST CODE	PARAMETERS

REQUEST CODE	FUNCTION
0000	Free Run
0001	Gain Setting For Tone
0010	Generate Tone
0011	Reserved
0100	Set ERL
0101	Reserved
0110	Set T/R & R/T ratios
0111	Set Tone Table *
1000	Set Speaker volume & line out gain
1001	Set Loop attenuation
1111	Read Internal Parameter (Reserved)
Others	Reserved

\* : Not available

**Request Code = 0000 (Free Run)**

	(15-12)	(11-8)	(7)	(6)	(5)	(4-0)
CONF	1100	0000	STOP	LINE MUTE	SPK MUTE	00000

STOP = 0 : turn on speakerphone  
       = 1 : turn off speaker phone  
 LINE MUTE = 1 : mute line out  
           = 0 : normal line out  
 SPK MUTE = 1 : mute loudspeaker  
           = 0 : normal speaker out

	(15-9)	(8)	(7-6)	(5)	(4-3)	(2-0)
RESP	0000000	CAS	00	TONE	00	LOOP ATTENUATION

TONE = 1 : Call Progress Tone is detected  
 Loop Attenuation : see Set Loop Attenuation  
 CAS : CAS detection (see mode 15)

**Request Code = 0001 (Set Gain for Tone)**

	(15-12)	(11-8)	(7-4)	(3-0)
CONF	1100	0001	GAIN-LO	GAIN-HI

GAIN-LO, GAIN-HI: 0~14 denote 0dB~-14dB; 15 denotes mute

	(15-3)	(2-0)
RESP	0000000000000	LOOP ATTENUATION

**GAIN TABLE:**

GAIN	0	1	2	3	4	5	6	7
dB	0dB	-1dB	-2dB	-3dB	-4dB	-5dB	-6dB	-7dB
GAIN	8	9	A	B	C	D	E	F
dB	-8dB	-9dB	-10dB	-11dB	-12dB	-13dB	-14dB	MUTE

**Request Code = 0010** (Send DTMF or Single Tone)

	(15-12)	(11-8)	(7)	(6)	(5)	(4-0)
CONF	1100	0010	START	0	D S	INDEX

START = 1 : start to generate Tone according to the index assigned

= 0 : stop generating Tone

D or S = 0 : generate Single Tone (This bit is valid under bit 4 set)

= 1 : generate Dual Tone

	(15-9)	(8)	(7-3)	(2-0)
RESP	0000000	CAS	00000	LOOP ATTENUATION

**INDEXTABLE:**

Index Value	Tone	Index Value	Tone
00000	DTMF_0	10000	
00001	DTMF_1	10001	
00010	DTMF_2	10010	
00011	DTMF_3	10011	
00100	DTMF_4	10100	
00101	DTMF_5	10101	
00110	DTMF_6	10110	
00111	DTMF_7	10111	
01000	DTMF_8	11000	
01001	DTMF_9	11001	
01010	DTMF_*	11010	
01011	DTMF_#	11011	
01100	DTMF_A	11100	
01101	DTMF_B	11101	
01110	DTMF_C	11110	
01111	DTMF_D	11111	

Note :

1. The tone frequency whose index is specified form 10000 to 11111 can be stored into the ROM space with starting address specified by the label "TONE\_TAB". At the same time, the label "TONE\_TAB" should be declared as global.
2. The legal index values for dual tone are just from 10000 to 10111. For dual tone mode, it will catch the frequency specified by both the index and index+8.

**Request Code = 0100 (Set ERL)**

	(15-12)	(11-8)	(7-4)	(3-0)
CONF	1100	0100	ERL_AEC	ERL_LEC

ERL\_AEC:echo return loss occurred on speaker side  
 ERL\_LEC:echo return loss occurred on line side

	(15-3)	(2-0)
RESP	0000000000000	LOOP ATTENUATION

ERL\_AEC & ERL\_LEC : Range from 0 to 30 with 2 dB/step

0000	00001	0010	0011	0100	0101	0110	0111
0 dB	2 dB	4 dB	6 dB	8 dB	10 dB	12 dB	14 dB
1000	1001	1010	1011	1100	1101	1110	1111
16 dB	18 dB	20 dB	22 dB	24 dB	26 dB	28 dB	30 dB

Note: The default value of ERL\_LEC and ERL\_AEC are 0 dB.

**Request Code = 0110 (Set T/R & R/T ratio)**

	(15-12)	(11-8)	(7-6)	(5-3)	(2-0)
CONF	1100	0110	00	R/T RATIO	T/R RATIO

	(15-3)	(2-0)
RESP	0000000000000	LOOP ATTENUATION

**T/R & R/T RATIO**

000	001	010	011	100	101	110	111
3 dB	6 dB	9 dB	12 dB	15 dB	Reserved	Reserved	Reserved

Note : 12 dB is default value

**Request Code = 0111 (Set Tone Table)**

	(15-12)	(11-8)	(7-4)	(3-0)
CONF	1100	0110	0000	INDEX

Index: Range is from 0 to 15 (see Request Code=0010)

RESP	(15-0) 0000000000000000
------	----------------------------

	(15-12)	(11-8)	(7-0)
CONF	1100	0111	FREQ_LO

RESP	(15-0) 0000000000000000
------	----------------------------

	(15-12)	(11-8)	(7-0)
CONF	1100	0111	FREQ_LO

FREQ\_HI and FREQ\_LO will compose a 16-bit frequency, which is equal to "target Frequency \*8.192".

RESP	(15-0) 0000000000000000
------	----------------------------

**Request Code = 1000 (Set SPK\_GAIN & LINE\_GAIN)**

	(15-12)	(11-8)	(7-4)	(3-0)
CONF	1100	1000	LINE_GAIN	SPK_GAIN

LINE\_GAIN:line out volume gain

SPK\_GAIN:speaker out volume gain

RESP	(15-3) 00000000000000	(2-0) LOOP ATTENUATION
------	--------------------------	------------------------------

SPK\_GAIN & LINE\_GAIN : Range from -20 to 20

0000	00001	0010	0011	0100	0101	0110	0111
-20dB	-16 dB	-12 dB	-8 dB	-4 dB	0 dB	2 dB	4 dB
1000	1001	1010	1011	1100	1101	1110	1111
6 dB	8 dB	10 dB	12 dB	14 dB	16 dB	18 dB	20 dB



**Request Code = 1001 (Set Loop Attenuation)**

	(15-12)	(11-8)	(7-3)	(2-0)
CONF	1100	1001	00000	LOOP ATTENUATION

	(15-3)	(2-0)
RESP	00000000000000	LOOP ATTENUATION

Loop Attenuation (CONF) : Range from 0 to 28 with 4dB/Step

000	001	010	011	100	101	110	111
0.dB	4 dB	8 dB	12 dB	16 dB	20 dB	24 dB	28 dB

Full Duplex

Note:8dB is default value

Smooth

Half Duplex

Loop Attenuation (RESP) :

000	001	010	011	100	101	110	111
0.dB	4 dB	8 dB	12 dB	16 dB	20 dB	24 dB	≥28 dB

**Request Code = 1111 (Read Internal Parameters)**

	(15-12)	(11-8)	(7-0)
CONF	1100	1111	Parameter Type

	(15-0)
RESP	Parameter

Parameter Type	Description
00000000	AEC_SNR1
00000001	LEC_SNR1
00000010	AEC_STATE
00000011	LEC_STATE
00000100	AEC_AERLE
00000101	LEC_AERLE
00000110	ACC_ATT
00000111	LEC_ATT
00001000	TR_MODE

**Read CODEC Register (Reference 93002)**

	(15-12)	(11-8)	(7)	(6-0)
CONF	1100	1111	1	Parameter Type

RESP	(15-0) Parameter
------	---------------------

CODEC Register	Description
0000000	Register 0
0000001	Register 1
0000010	Register 2
0000011	Register 3
0000100	Register 4
0000101	Register 5
0000110	Register 6
0000111	Register 7

**Mode 13. PERSONAL MAIL MODE**

	(15-12)	(11-8)	(7-5)	(4)	(3)	(2-0)
CONF	1101	0000	000	LOOP BACK	MEMO /ICM	PERSONAL MAIL

PERSONAL MAIL : 0 ~ 7, total 8 personal mailboxes. The MSG IDs of PLAY MODE, DELETE MESSAGE MODE and GET RECORD MESSAGE INFORMATION MODE are all limited to the personal mailbox specified by this command.

MEMO/ICM = 0 : set the attribute of the recorded message to be ICM.  
 = 1 : set the attribute of the recorded message to be MEMO.

LOOP BACK = 0 : enable loop back.  
 = 1 : disable loop back. Then in RECORD and LINE MONITOR modes, the codec-in data will not be put on the codec-out path.

RESP	(15-0) 0000000000000000
------	----------------------------

**Set Some Status,**

	(15-12)	(11-8)	(7)	(6)	(5-3)	(3)	(2)	(0)
CONF	1101	0001	CAS SPK	0	COMPRESS FORMAT	SILENCE RATE	CODEC FORMAT	OUTPUT LEVEL

**OUTPUT LEVEL**

(set the relation of the output speech level and the corresponding input speech level)

= 0 : the output level is 5dB less than the input level (default setting).

= 1 : the output level (playback) is equal to the input level (recording).

**CODEC FORMAT** (set the Codec data format)

= 0 : 8-bit m-law data format is selected.

= 1 : 16-bit linear data format is selected.

(Note: Macronix MX93000 Codec family provide this mode.)

**SILENCE RATE**

(set the silence storage rate, it is significant only when COMPRESS FORMAT = 000)

= 0 : silence storage rate is 0.8kbps. (default setting)

= 1 : silence storage rate is 4.8kbps.

**COMPRESS FORMAT** (select the speech compression algorithm for message recording)

= 000 : CELP 4.8kbps with silence compression.

= 001 : CELP 12.8kbps with silence suppression.

= 010 - 111 : reserved.

**CAS SPK**

= 0 : turn off detecting CAS within speaker phone

= 1 : turn on detecting CAS within speaker phone

	(15-0)
RESP	0000000000000000

**Set Telephone Data Configuration,**

	(15-12)	(11-8)	(7-5)	(4-2)	(1)	(0)
CONF	1101	0011	000	TEL RECORD NUMBER	0	1

**TEL RECORD NUMBER** : set the number of total telephone data records,

= 000 : 0 record (default setting).

= 001 : 63 records.

= 010 : 126 records.

= 011 : 189 records.

= 100 : 252 records.

	(15-0)
RESP	0000000000000000

**Voice Activity Mode,**

	(15-12)	(11-8)	(7-6)	(5-0)
CONF	1101	0111	00	VOX Level

VOX LEVEL : see Table 6.

	(15-0)
RESP	0000000000000000

**Mode 14. TELEPHONE DATA MODE**
**WRITE TELEPHONE DATA**

	(15-12)	(11-10)	(9)	(8)	(7-0)
CONF (C1)	1110	00	STOP	0	TEL RECORD ID

TEL RECORD ID:1~255, each telephone record has maximum 60 bytes.

STOP =1:stop telephone data writing.

	(15-12)	(11-10)	(9-1)	(0)
RESP (R1)	1110	00	00000000	FULL

FULL=1: no free telephone record is available.

	(15-12)	(11-10)	(9)	(8)	(7-0)
CONF (C2)	1110	00	STOP	0	DATA

DATA : the telephone data to write.

STOP =1:stop telephone data writing.

	(15-12)	(11-10)	(9-1)	(0)
RESP (R2)	1110	00	00000000	FULL

FULL =1:current telephone record is full.

**READ TELEPHONE DATA**

	(15-12)	(11-10)	(9)	(8)	(7-0)
CONF	1110	01	STOP	0	TEL RECORD ID

TEL RECORD ID : 1 ~ 255.

STOP = 1 : stop telephone data reading.

	(15-12)	(11-10)	(9)	(8)	(7-0)
RESP	1110	00	0	READ END	DATA

DATA : the read out telephone data.

READ END = 1 : end of the current telephone record and the followed DATA is invalid.

**Mode 15. CALLER\_ID MODE**

	(15-12)	(11-9)	(8-0)
CONF	1111	REQUEST CODE	PARAMETER

REQUEST CODE	FUNCTION
000	Demodulate FSK
100	Set Response method of CAS detected
101	Set CAS sensitivity
001	ACK DTMF A
010	ACK DTMF D

**Request Code = 000 (Demodulate FSK To RAW DATA)**

	(15-12)	(11-8)	(8)	(7-2)	(1)	(0)
CONF	1111	000	0	000000	TYPE	START\

TYPE = 0 : CID TYPE I 200 bits channel seizure &amp; 10 bits mark signal

= 1 : CID TYPE II 10 bits mark signal

START\ = 0 : Start / Continue to Demodulate

= 1 : Stop Demodulating

	(15)	(14)	(13)	(12-8)	(7-0)
RESP	CD	ERR	DR	00000	FSK DATA BYTE

- CD = 0 : Carrier is not detected (FSK signal do not occur on line)  
 = 1 : Carrier is detected (FSK signal takes place on line)
- ERR = 0 : Data buffer is not overflow  
 = 1 : Data buffer is overflow
- DR = 0 : FSK data byte is not ready  
 = 1 : FSK data byte is ready

Note : The length for RAW Data buffer is 8 bytes long.

**Request Code = 000 (Demodulate FSK To COOKED DATA)**

	(15-12)	(11-8)	(8)	(7-2)	(1)	(0)
CONF	1111	000	1	000000	TYPE	START\

- TYPE = 0 : CID TYPE I 200 bits channel seizure & 10 bits mark signal  
 = 1 : CID TYPE II 10 bits mark signal
- START\ = 0 : Start / Continue to Demodulate  
 = 1 : Stop Demodulating

	(15)	(14)	(13)	(12-8)	(7-0)
RESP	CD	ERR	DR	00000	FSK DATA BYTE

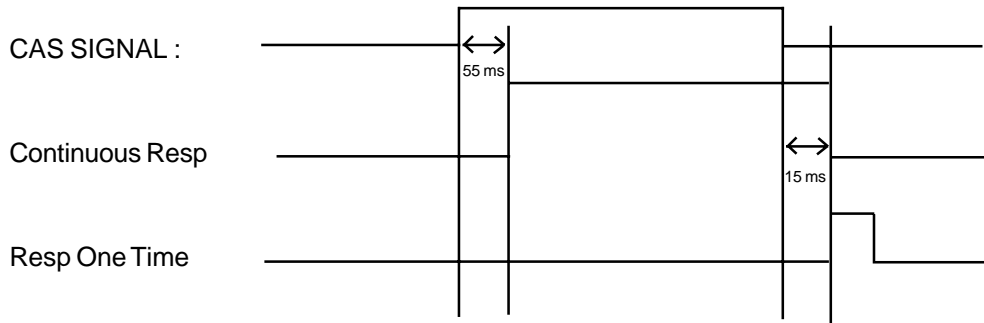
- CD = 0 : Carrier is not detected (FSK signal do not occur on line)  
 = 1 : Carrier is detected (FSK signal takes place on line)
- ERR = 0 : Data buffer is not overflow  
 = 1 : Data buffer is overflow
- DR = 0 : FSK data byte is not ready  
 = 1 : FSK data byte is ready

- Note:** 1) The second byte data received by FSK decoder will be considered as the total data length.  
 2) The first byte sent out are the total length +1 in which the checksum is included.  
 3) The length for cooked data buffer is 60 bytes long.

**Request Code = 100 (Set Response method of CAS detected)**

	(15-12)	(11-9)	(8-1)	(0)
CONF	1111	000	00000000	CAS TYPE

CAS TYPE = 0 : Continuous Response. Response CAS behavior fidelity (Delay about 55ms)  
 = 1 : Response One Time. Response CAS detected after CAS is off



	(15-0)
RESP	0000000000000000

**Request Code = 101 (Set CAS Sensitivity)**

	(15-12)	(11-9)	(8-3)	(2-0)
CONF	1111	101	000000	CAS_LVL

	(15-0)
RESP	0000000000000000

	0	1	2	3	4	5	6	7
Sensitivity	-32	-34	-36	-38	-40	-42	-44	-46

**Request Code = 001 or 010 (ACK DTMF A or ACK DTMF D)**

	(15-12)	(11-9)	(8)	(7-4)	(3-0)
CONF	1111	ACK	TO	GAIN2	GAIN1

GAIN1, GAIN2 : 0~14 denote 3dB ~ -25dB (2dB/one step), 15 denotes no related signal.

TO = 0 OUTPUT CHANNEL CODEC 0.

= 1 OUTPUT CHANNEL CODEC 1.

ACK =001 ACK DTMF A.

=010 ACK DTMF D.

	(15-0)
RESP	0000000000000000

**TABLE 1. PERFORMANCE TEST FOR TONE DETECTION**

(noiseless) Power (dB)	0	-3	-6	-10	-20	-30	-33	-36	-39
p2p (volt)	5	3.54	2.5	1.58	.5	.158	.112	.079	.056
280 Hz	x	x	x	x	x	x	x	x	x
300 Hz	ok	ok	ok	ok	ok	ok	ok	x	x
400 Hz	ok	ok	ok	ok	ok	ok	ok	ok	x
500 Hz	ok	ok	ok	ok	ok	ok	ok	ok	x
600 Hz	ok	ok	ok	ok	ok	ok	ok	ok	x
640 Hz	ok	ok	ok	ok	ok	ok	ok	x	x
670 Hz	x	x	x	x	x	x	x	x	x
480+620 Hz	ok	ok	ok	ok	ok	ok	x	x	x
350+440 Hz	ok	ok	ok	ok	ok	ok	x	x	x

\* 0 dB is defined as the Vrms of maximum codec input

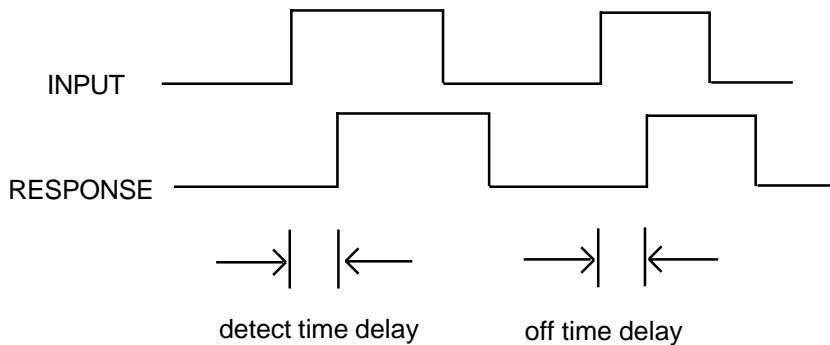
\*\* The maximum peak-to-peak voltage (P2P) of Codec input is assumed to be 5V



SNR (dB)	40	25	12
detect time delay / off time delay (ms)			
300 Hz	50 / 40	55 / 40	65 / 40
400 Hz	40 / 40	40 / 40	45 / 45
500 Hz	40 / 40	40 / 40	45 / 45
640 Hz	45 / 40	45 / 40	60 / 35
480+620 Hz	40 / 40	45 / 40	50 / 40
350+440 Hz	45 / 40	45 / 40	55 / 40

\* 0 dB is defined as the  $V_{rms}$  of maximum codec input

\*\* the detect/off time delay is defined as follows



**TABLE 2. RESOLUTION TEST FOR DUAL TONE (FREQUENCY DETECTABLE RANGE)**

RANGE / POWER	-4 dB	-25 dB
(300 + F2 Hz)	F2 > 320 Hz	F2 > 320 Hz
(640 + F2 Hz)	F2 < 620 Hz	F2 < 620 Hz

**TABLE 3. THD DISTORTION TEST FOR DUAL TONE (DISTORTION TOLERANCE)**

INPUT / POWER	-4 dB	-25 dB
(300 + 320 Hz)	> 10 dB	> 10 dB
(640 + 620 Hz)	> 10 dB	> 10 dB

\* 0 dB is defined as the  $V_{rms}$  of maximum codec input

**TABLE 4. CALL PROGRESS TONE DETECTOR CHARACTERISTICS**

PARAMETER	MIN	TYP	MAX	UNIT
Detection level *	-30			dB
Rejection level			-37	dB
Frequency range	300		640	Hz
Noise tolerance			-12	dB
THD tolerance			-10	dB
Rejection frequency range	< 280 Hz or		> 670 Hz	
Minimum signal duration accept	50 ms			
Maximum response time	50 ms			

\* 0 dB is defined as the Vrms of maximum codec input

**TABLE 5. DTMF CHARACTERISTICS**

PARAMETER	MIN	TYP	MAX	UNIT
DTMF signal level for detection *	-43		0	dB
DTMF twist (high/low tone)		± 9		dB
DTMF noise tolerance		-10		dB
DTMF tone duration accept	50			ms
DTMF interdigit pause accept	50			ms
DTMF frequency deviation accept limit		±1.5		%
DTMF frequency deviation reject limit		± 3		%
Tone generator level	-25		+3	dB
Tone generator frequency accuracy		±0.1		%
Tone generator level accuracy		±0.5		dB

\* 0 dB is defined as the Vrms of maximum codec input

**TABLE 6. VOX THRESHOLD**

dB	VOX_LEVEL
-9	0
-10	1
-11	2
-12	3
-13	4
-14	5
-15	6
-16	7
-17	8
-18	9
-19	10
-20	11
-21	12
-22	13
-23	14
-24	15
-25	16
-26	17
-27	18
-28	19
-29	20
-30	21
-31	22
-32	23
-33	24
-34	25
-35	26
-36	27
-37	28
-38	29
-39	30
-40	31
-41	32
-42	33
-43	34
-44	35
-45	36
-46	37
-47	38
-48	39
-49	40
-50	41
-51	42
-52	43
-53	44
-54	45
-55	46
-56	47
-57	48
-58	49
-59	50
-60	51
-61	52
-62	53

0 dB is defined as the Vrms of maximum Codec input.

**TABLE 7. SILENCE THRESHOLD LEVEL**

SILENCE LEVEL	dB
1	-58
2	-55
3	-52
4	-49
5	-46
6	-43
7	-40
8	-37
9	-34
10	-31
11	-28
12	-25
13	-22
14	-19
15	-16

0 dB : maximum input of sinusoidal wave

**TABLE 8. ECHO CANCELLER PERFORMANCE \***

Test Description	
Final Echo Return Loss Improvement (ERLI) (Singletalk mode)	> 28 dB
Maximum tail length	12 ms
Convergence Rate (Singletalk mode) 500 ms after initialization with cleared register and with near-end signal set to zero at initialization time	> 28 dB
Infinite Return Loss Convergence Return Echo Loss Improvement 500 ms after echo path is interrupted	> 28 dB

\* test signal : band-limited white-noise (300-3400 Hz)

**TABLE 9. MAXIMUM COMMAND RESPONSE TIME**

COMMAND	MAX RESP TIME (ms)
Record Mode :	
1. start	32
2. normal	6
3. stop	104
Play Mode :	
1. start	14
2. normal	5
3. stop	1
Message Status Mode :	
1. get message number	8
2. get FLASH available time	6
3. get FLASH garbage collection recommendation status	1
DTMF Generator Mode	1
Line Monitor Mode	6
Delete Message Mode :	
1. delete one message	4300
2. delete all old messages	5800
3. real delete	5500
Set Current Time/Voice Prompt/Silence Threshold Mode	1
Get Current Time Mode	1
Initial and Test Memory Mode :	
1. FLASH re-initialization	2400
2. FLASH initialization/error correction	5800
3. FLASH garbage collection	150
4. other misc. setting	1
Get Record Message Information Mode	10
Voice Prompt Mode :	
1. start	6
2. normal	5
3. stop	1
Voice Activity Mode	1
Personal Mail Mode	1

**APPENDIX A : CID TEST REPORT BASED ON SR-3004**

## 1. Type 1 Testing

## 1.1 Testing Items

A list of parameters to test based on SR-3004 are divided into two categories: Transmission tests and Message Format tests. Transmission tests vary electrical properties of signals to identify when and where CPE fails to detect. The Message Format tests vary the content and formatting of the data message. The following tests assumes that Transmission testing is independent of Message Format testing. The FSK mode provided on the MX93032-M1 is just responsible to decode the physical layer and data link layer. Therefore, Only transmission test associated with the MX93032-M1 will be given. For message format tests, it should be tested by "system manufacturers" depending on the their interesting region.

## 1.2 Transmission Testing

Mark Frequency  
 Space Frequency  
 Twist Immunity  
 Noise Immunity  
 Baud Rate

## 1.2.1 Immunity to Frequency and Baud Rate Variation

Item	Testing Range	Desired Range
SPACE	2100HZ ~ 2300HZ	2178HZ ~ 2222HZ
MARK	1100HZ ~ 1300HZ	1188HZ ~ 1212HZ
BAUD RATE	1150 baud ~ 1250 baud	1188 baud ~ 1212 baud

## Testing Condition :

Signal amplitude at -20dBm for per frequency

SNR : 40 dB

Signal : For the signal which meets the following each conditions will be send 100 times, the times CPE receive correctly will be recorded.

## Baud Rate = 1188 baud

	2100	2150	2178	2222	2250	2300
1100	95	100	100	100	100	100
1150	100	100	100	100	100	100
1188	100	100	100	100	100	100
1212	100	100	100	100	100	100
1250	100	100	100	100	100	100
1300	100	100	100	100	100	80

Baud Rate = 1212 baud

	2100	2150	2178	2222	2250	2300
1100	100	100	100	100	100	100
1150	100	100	100	100	100	100
1188	100	100	100	100	100	100
1212	100	100	100	100	100	100
1250	100	100	100	100	100	100
1300	100	100	100	100	100	100

### 1.2.2 Twist Immunity

Range +/- 10 dB

Fix Condition:

Space Frequency : 2200 HZ

Mark Frequency : 1200 HZ

Baud Rate : 1200 Baud

SNR : 40 dB

Signal : For the signal which meets the following each conditions will be send 100 times, the times CPE receive correctly will be recorded.

Positive Twist : (Mark > Space)

Mark is fixed to -13 dBm and Space vary from -13 dBm to -48 dBm with step 5 dB

-13	-18	-23	-28	-33	-38	-43	-48
100	100	100	0	0	0	0	0

Negative Twist : (Mark < Space)

Space is Fixed to -13 dBm and Mark vary from -13dBm to -48 dBm with step 5 dB

-13	-18	-23	-28	-33	-38	-43	-48
100	100	100	0	0	0	0	0

### 1.3.4 Noise Immunity:

Fix Condition:

Space Frequency : 2200 HZ

Mark Frequency : 1200 HZ

Baud Rate : 1200 Baud

Signal Level : -20 dBm

Signal : For the signal which meets the following each conditions will be send 100 times, the times CPE receive correctly will be recorded.

10	12	14	15	18	20	22	24
0	0	0	100	100	100	100	100

26	28	30	32	34	36	38	40
100	100	100	100	100	100	100	100

## TYPE II TESTING

The Type 2 CPE requires all of the same conformance test as Type 1 CPE in addition to "CAS detector performance"

## 2.1 CAS Detector Performance

There are three types of tests to perform for CAS detector performance:

Signal Recognition Tolerance ( without speech )

Talk off ( false detection due to speech )

Talk Down (missed detection due to interference from speech )

## 2.1.1 Signal Recognition Tolerance

Parameter	Nominal	Minimum	Maximum
Lower Frequency	2130 +/- 2 Hz	2119 Hz	2141 Hz
Upper Frequency	2750 +/- 2 Hz	2736 Hz	2764 Hz
Power level	-22 +/- 1 dBm	-32 dBm	-14 dBm
Twist	0 ~ 2 dB	0 dB	6 dB
Duration	80 +/- 1 ms	75 ms	85 ms

Various combinations of the above parameter values will be tested by sending 1000 CAS tones to the CPE at a rate not exceeding 2 per second. The various parameter combinations are broken into three groups described as follows. Each test combination requires 34 minutes to complete with 1000 burst at a 2 second interval.

Group 1 consists of all the parameters at their nominal value.

Group 2 sets each parameter ( one at a time) to its extreme values, while the remaining parameters remain at their nominal value.

Group 3 sets all parameters to 90% of its extreme value. All of the possible parameter combinations are tested.



**Group 1 :**

Test No	Frequency ( Low , High)	Power Level (Low , High)	Duration	Correct Times
1	(2130,2750)	(-22,-22)	80	1000

**Group 2 :**

Test No	Frequency ( Low , High)	Power Level (Low , High)	Duration	Correct Times
1	(2119,2750)	(-22,-22)	80	1000
2	(2141,2750)	(-22,-22)	80	1000
3	(2130,2736)	(-22,-22)	80	1000
4	(2130,2764)	(-22,-22)	80	1000
5	(2130,2750)	(-14,-14)	80	1000
6	(2130,2750)	(-32,-32)	80	1000
7	(2130,2750)	(-14,-20)	80	1000
8	(2130,275)	(-32,-26)	80	1000
9	(2130,2750)	(-20,-14)	80	1000
10	(2130,2750)	(-26,-32)	80	1000
11	(2130,2750)	(-22,-22)	75	1000
12	(2130,2750)	(-22,-22)	85	1000



**Group 3 :**

Test No	Frequency ( Low , High)	Power Level (Low , High)	Duration	Correct Times
1	(2120,2737)	(-31,-31)	76	1000
2	(2120,2737)	(-31,-26)	76	1000
3	(2120,2737)	(-26,-31)	76	1000
4	(2120,2737)	(-15,-15)	76	1000
5	(2120,2737)	(-15,-20)	76	1000
6	(2120,2737)	(-20,-15)	76	1000
7	(2120,2737)	(-31,-31)	84	1000
8	(2120,2737)	(-31,-26)	84	1000
9	(2120,2737)	(-26,-31)	84	1000
10	(2120,2737)	(-15,-15)	84	1000
11	(2120,2737)	(-15,-20)	84	1000
12	(2120,2737)	(-20,-15)	84	1000
13	(2140,2737)	(-31,-31)	76	1000
14	(2140,2737)	(-31,-26)	76	1000
15	(2140,2737)	(-26,-31)	76	1000
16	(2140,2737)	(-15,-15)	76	1000
17	(2140,2737)	(-15,-20)	76	1000
18	(2140,2737)	(-20,-15)	76	1000
19	(2140,2737)	(-31,-31)	84	1000
20	(2140,2737)	(-31,-26)	84	1000
21	(2140,2737)	(-26,-31)	84	1000
22	(2140,2737)	(-15,-15)	84	1000
23	(2140,2737)	(-15,-20)	84	1000
24	(2140,2737)	(-20,-15)	84	1000
25	(2120,2763)	(-31,-31)	76	1000
26	(2120,2763)	(-31,-26)	76	1000
27	(2120,2763)	(-26,-31)	76	1000
28	(2120,2763)	(-15,-15)	76	1000
29	(2120,2763)	(-15,-20)	76	1000
30	(2120,2763)	(-20,-15)	76	1000
31	(2120,2763)	(-31,-31)	84	1000
32	(2120,2763)	(-31,-26)	84	1000
33	(2120,2763)	(-26,-31)	84	1000
34	(2120,2763)	(-15,-15)	84	1000
35	(2120,2763)	(-15,-20)	84	1000
36	(2120,2763)	(-20,-15)	84	1000
37	(2140,2763)	(-31,-31)	76	1000
38	(2140,2763)	(-31,-26)	76	1000
39	(2140,2763)	(-26,-31)	76	1000
40	(2140,2763)	(-15,-15)	76	1000
41	(2140,2763)	(-15,-20)	76	1000
42	(2140,2763)	(-20,-15)	76	1000
43	(2140,2763)	(-31,-31)	84	1000
44	(2140,2763)	(-31,-26)	84	1000
45	(2140,2763)	(-26,-31)	84	1000
46	(2140,2763)	(-15,-15)	84	1000
47	(2140,2763)	(-15,-20)	84	1000
48	(2140,2763)	(-20,-15)	84	1000

**CAS Level Reject Limit**

Test No	Frequency ( Low , High)	Power Level (Low , High)	Duration	Correct Times
1	(2120,2750)	(-46,-46)	80	0

**2.1.1.2 Talk-down Test**

Row	Signal Level (dBm/Tone)	Speech Level (dBm)	SNR	CAS Pulses Received	CAS Pulses Sent	Derated Miss Rate (%)
1	-28	-10	-18	3608	6000	19.933
2	-28	-13	-15	4586	6000	11.7833
3	-22	-10	-12	909	1125	9.6
4	-26	-16	-10	1073	1125	2.311
5	-28	-22	-6	1112	1125	0.5778
6	-22	-19	-3	1124	1125	0.0444

**CAS LEVEL**

Speech	-30 dBm	-28 dBm	-26 dBm	-24 dBm	-22 dBm	-20 dBm	-18 dBm	-16 dBm
-7 dBm	33.5166^	28.0833^	22.6500^	17.2166~	11.7833	10.3278~	5.9556~	1.8778~
-10 dBm	25.3666^	19.933	14.5000~	11.0555~	9.60	2.3111	1.4445~	0.5778
-13 dBm	17.2166~	11.7833	10.3278~	5.9556~	1.8778~	1.0111~	0.4000~	0.0444
-16 dBm	11.0555~	9.6	2.3111	1.4445~	0.5778	0.2222~	0*	0*
-19 dBm	5.9556~	1.8778~	1.0111~	0.4~	0.0444	0*	0*	0*
-22 dBm	1.4445~	0.5788	0.2222~	0*	0*	0*	0*	0*
-25 dBm	0.4~	0.0444	0*	0*	0*	0*	0*	0*
-28 dBm	0*	0*	0*	0*	0*	0*	0*	0*
-31 dBm	0*	0*	0*	0*	0	0*	0*	0*

in which the symbols "~" , "^" and "\*" represent interpolated, extrapolated and clamped to minimum or maximum, respectively.

Weighted average detection rate for all 72 level combinations given above

	-30	-28	-26	-24	-22	-20	-18	-16	Sum
-7	0.000129	0.000172	0.000344	0.000731	0.000731	0.000731	0.000731	0.000731	0.0043
-10	0.000774	0.001032	0.002064	0.004386	0.004386	0.004386	0.004386	0.004386	0.0258
-13	0.002973	0.003964	0.007928	0.016847	0.016847	0.016847	0.016847	0.016847	0.0991
-16	0.006684	0.008912	0.017824	0.037876	0.037876	0.037876	0.037876	0.037876	0.2228
-19	0.0088	0.01184	0.02368	0.05032	0.05032	0.05032	0.05032	0.05032	0.296
-22	0.006684	0.008912	0.017824	0.037876	0.037876	0.037876	0.037876	0.037876	0.2228
-25	0.002973	0.003964	0.007928	0.016847	0.016847	0.016847	0.016847	0.016847	0.0991
-28	0.000774	0.001032	0.002064	0.004388	0.004386	0.004386	0.004386	0.004386	0.0258
-31	0.000129	0.000172	0.000344	0.000731	0.000731	0.000731	0.000731	0.000731	0.0043
Sum	0.03	0.04	0.08	0.17	0.17	0.17	0.17	0.17	1

Talkdown Condition 1 -- the combination of average signal level (-22 dBm /tone) and average speech level (-19 dBm). At least 99.5% of the signals sent should be received correctly using derated missed rate.

Talkdown Condition 2 -- the weight average of the worst one percent of the 72 combinations. At least 93% of the signal sent should be received correctly using the derated missed rate.

Talkdown Condition 3 -- the weighted average of all of the 72 combinations. At least 99.5% of the signal should be received correctly using derated missed rate.\

Statistical Talkdown Result:

Condition 1 (nominal) : 99.956 % ( Spec. >= 99.5% )  
 Condition 2 (worst 1 %) : 83.784 % ( Spec. >= 93 % )  
 Condition 3 (global) : 99.005 % ( Spec. >= 99.5%)

**2.1.1.3 Talk-off Test**
**Talkoff Test Raw Data:**

Row	Near End Speech Level (dBm)	Hits No on Near End Speech	Testing Time	Far End Speech Level (dBm)	Hits No on Far End Speech	Testing Time
1	-10	8	96			
2	-16	5	96	-16	5	96
3	-22	3	96	-22	3	96

**FAR END SPEECH**

	-13 dBm <sup>^</sup>	-16 dBm	-19 dBm <sup>~</sup>	-22 dBm	-25 dBm <sup>^</sup>	-28 dBm <sup>^</sup>	-31 dBm <sup>^</sup>
<b>N</b>							
<b>E</b>	-7 dBm <sup>^</sup>	0.0729	0.0677	0.0625	0.0573	0.0521	0.0469
<b>A</b>							
<b>R</b>	-10 dBm	0.0729	0.0677	0.0625	0.0573	0.0521	0.0469
<b>E</b>	-13 dBm <sup>~</sup>	0.0651	0.0599	0.0547	0.0495	0.0443	0.0391
<b>N</b>	-16 dBm	0.0573	0.0521	0.0469	0.0417	0.0365	0.0313
<b>D</b>	-19 dBm <sup>~</sup>	0.0521	0.0469	0.0417	0.0365	0.0313	0.0260
<b>S</b>	-22 dBm	0.0469	0.0417	0.0365	0.0313	0.0260	0.0208
<b>P</b>	-25 dBm <sup>^</sup>	0.0417	0.0365	0.0313	0.0260	0.0208	0.0156
<b>E</b>	-28 dBm <sup>^</sup>	0.0365	0.0313	0.0260	0.0208	0.0156	0.0104
<b>C</b>							
<b>H</b>	-31 dBm <sup>^</sup>	0.0313	0.0260	0.0208	0.0156	0.0104	0.0052
							0

~ interpolated    ^ extrapolated

**Weighted average of all 63 combinations**

	-13	-16	-19	-22	-25	-28	-31	Sum
-7	3.07E-5	9.013E-5	0.0002528	0.0005126	0.0008063	0.000915	0.0016925	0.0043
-10	0.0001842	0.0005408	0.001517	0.0030754	0.0048375	0.0054902	0.0101549	0.0258
-13	0.0007076	0.0020771	0.0058271	0.0118127	0.0185813	0.0210885	0.0390058	0.0991
-16	0.0015908	0.0046699	0.0131006	0.0265578	0.041775	0.0474118	0.0876941	0.2228
-19	0.0021134	0.0062042	0.0174048	0.0352832	0.0555	0.0629888	0.1165056	0.296
-22	0.0015908	0.0046699	0.0131006	0.0265578	0.041775	0.0474118	0.0876941	0.2228
-25	0.0007076	0.0020771	0.0058271	0.0118127	0.0185813	0.0210885	0.0390058	0.0991
-28	0.0001842	0.0005408	0.001517	0.0030754	0.0048375	0.0054902	0.0101549	0.0258
-31	3.07E-5	9.013E-5	0.0002528	0.0005126	0.0008063	0.000915	0.0016925	0.0043
Sum	0.00714	0.02096	0.0588	0.1192	0.1875	0.2128	0.3935	1

Talkoff Condition 1 : the combination of average near-end level (-19 dBm) and the average far end level (-28 dBm).

This talkoff rate should be less than 1 in 45 hours

Talkoff Condition 2 : the weighted average of the worst one percent of the 63 combinations. The talkoff rate should be less than 1 in 10 hours.

Talkoff Condition 3 : the weighted average of all of the 63 combinations. The talk off rate should be less than 1 in 35 hours.

**Tested Result :**

- Condition 1 (nominal) : 1 hit per 38.4 hours ( Spec. >= 45 hours)
- Condition 2 (worst 1 %) : 1 hit per 16.79 hours ( Spec. >= 10 hours)
- Condition 3 (global) : 1 hit per 35.73 hours ( Spec. >= 35 hours)

**3. Recommendation:**

All of the test results corresponding to talkdown given above are obtained by configuring our demo board to "Adjunct Model". That is, our demo board was connected directly to the Tip and Ring on which the far end speech and far end CAS has been mixed. Therefore, there is no way to tell speech signal from the CAS on that testing configuration adopted by us. For practical application, Near end speech instead of far end speech should occur on talkdown test. If the mixed signal captured from tip and ring can be subtracted from the source signal captured from the near end speech with proper gain. Then the SNR of resultant signal will be highly improved on adjusting the impedance match between CPE and SPCS adequately. According to our experience, if the quantity of the echo canceller can excess 6 dB, the performance of CAS detector embedded on 93032-M1 will meet the Bellcore specification for talkdown. The feature of echo canceller will be inherently possessed by phone mode, there will be additional circuit to implement echo canceller for adjunct model.

**APPENDIX B : OPERATION GUIDE FOR THE 93032-M1 DEMO BOARD**

## 1. INTRODUCTION:

The evaluation board (EVB) is provided by Macronix is to demonstrate both Digital speakerphone, Caller ID and DAM functions. There are four parts in EVB:

1. Main Module with label "A001-BD-0038A"
2. DSP Module with label "A001-BD-0031B"
3. DAA Module with label "MSP-PCB-010A"
4. MCU Module with label "A001-BD-0040A"

The key pad is arranged into six kinds of definitions:

1. DAM mode keys: The power on default key function to demo DAM record/play/etc functions.
2. Speakerphone mode keys: Press "spk phone on" in DAM mode to enter speakerphone mode. User can dial DTMF, set loudspeaker / lineout volume or mute line-out signal.
3. Speaker test mode keys : Press "test mode" key in speakerphone mode for 1 second to enter this mode which allows user to set all the speakerphone BIOS parameters through key pad.
4. CPE Alerting Signal (CAS) test : Press "CAS Test" and last for 1 second in DAM mode to enter CAS test mode. During this mode , the number of CAS signal received from Tip and Ring will be counted and represented on LCD display.
5. Caller ID TypeII Test mode: Press "CAS Test" within 1 second in DAM mode to come into Caller ID Type II Test mode. After coming into this mode, 93032-M1 will monitor line and detect the CAS signal presented on line. The DTMF will be sent out as the CAS was found, following the FSK will be decoded and showed on LCD.
6. Encoding mode selection : Press "Enc. Select" in DAM mode to switch the encoding mode either 4.8 k or 12.8 k.
7. Flash Re-initialization : Press "Flash Re-Init." in DAM mode to perform FLAH Re-Initialization

**2. MATRIX KEY DEFINITION:**

Key Pad under DAM mode (Power on default mode)

ANN1 1	ON/OFF 2		Flash Re-Init. A
ANN2 4	PLAY 5		CAS TesT B
ANN3 7	DEL 8		Enc. Select C
CODE/FF 0	TIME/ STOP F	MEMO/ REV E	SPK.Phone ON/OFF D

Key Pad in Speaker Phone mode :

1 1	2 2	3 3	Spk Vol + A
4 4	5 5	6 6	Spk Vol - B
7 7	8 8	9 9	Mute Lout/ Test Mode C
Lout Vol + 0	0 F	Lout Vol - E	SPK.Phone OFF D



## Key Pad in Speaker Phone Test mode :

Current Loop Att	Set LINE_DRV	Set T/R Ratio	Target Loop Att +
1	2	3	A
	Set AD2_PGA	Set R/T Ratio	Target Loop Att -
4	5	6	B
Set Pre_PGA	Set ATT1	Set LEC_ERL	Exit Test Mode
7	8	9	C
Set AD1_PGA	Set SPK_DRV	Set AEC_ERL	
0	F	E	D

## Key Pad in CALLER ID Test mode :

		R/C DUMP CID	
1	2	3	A
			CALLER TYPE II/CAS TEST
4	5	6	B
7	8	9	C
Forward Caller ID Data	STOP	Backward Caller ID Data	
0	F	E	D

**Key Pad Definitions under DAM/Caller ID mode:**

NO	FUNCTION	PRESS METHOD
1	OGM1 PLAY	Press within 2 sec
	OGM1 RECORD AND PLAY	Press over 2 sec
2	Turn on DAM	Press
3	Raw/Cook Data Select	Press within 2 sec
	Dump Caller ID Data	Press over 2 sec
4	OGM2 PLAY	Press within 2 sec
	OGM2 RECORD AND PLAY	Press over 2 sec
5	PLAY ALL MESSAGE	Press within 2 sec
	PLAY NEW MESSAGE	Press over 2 sec
6		
7		
8	Delete Current Message played	Press
9		
0	Show And Announce Password	Press within 2 sec
	Set Password	Press over 2 sec
	Forward (for playing message and setting password)	Press
A	Flash Re-initialization	Press
B	CAS Test	Press
C	Enc. Selection	Press
E	Record Memo	Press over 2 sec
	Rewind (for playing message and setting password)	Press
F	Announce Time Stamp	Press
	Stop	Press

**Key Pad Definitions under Speaker Phone Mode :**

NO	FUNCTION	PRESS METHOD
1	GENERATE DTMF "1"	Press
2	GENERATE DTMF "2"	Press
3	GENERATE DTMF "3"	Press
4	GENERATE DTMF "4"	Press
5	GENERATE DTMF "5"	Press
6	GENERATE DTMF "6"	Press
7	GENERATE DTMF "7"	Press
8	GENERATE DTMF "8"	Press
9	GENERATE DTMF "9"	Press
0	Increase Line Out Volume	Press
A	Increase Speaker Volume	Press
B	Decrease Speaker Volume	Press
C	Mute Line Out	Press within 2 sec
	Enter Test Mode	Press over 2 sec
D	Turn Off Speaker Phone	Press
E	Decrease Line Out Volume	Press
F	GENERATE DTMF "0"	Press

**Key Pad Definitions under Speaker Phone Mode (Test Mode) :**

NO	FUNCTION	PRESS METHOD
1	Show Current Loop Attenuation	Press
2	SET LINE_DRV (MX93002 CODEC)	Press
3	SET T/R RATIO	Press
4		
5	SET AD2_PGA (MX93002 CODEC)	Press
6	SET R/T RATIO	Press
7	SET PRG_PGA (MX93002 CODEC)	Press
8	SET_ATT1(MX93002 CODEC)	Press
9	SEL LEC_ERL	Press
0	SET AD1_PGA	Press
A	TARGET LOOP ATTENUATION UP	Press
B	TARGET LOOP ATTENUATION DOWN	Press
C	EXIT TEST MODE	Press
D		
E	SET AEC_ERL	Press
F	SET SPK_DRV(MX93002 CODEC)	Press

**Speaker Phone related BIOS parameters**

ERL\_AEC & ERL\_LEC : Range from 0 to 30 with 2 dB/step

0000	00001	0010	0011	0100	0101	0110	0111
0 dB	2 dB	4 dB	6 dB	8 dB	10 dB	12 dB	14 dB
1000	1001	1010	1011	1100	1101	1110	1111
16 dB	18 dB	20 dB	22 dB	24 dB	26 dB	28 dB	30 dB

Note: The default value of ERL\_LEC and ERL\_AEC are 0 dB.

T/R & R/T Ratio

000	001	010	011	100	101	110	111
3 dB	6 dB	9 dB	12 dB	15 dB	Reserved	Reserved	Reserved

Note : 12 dB is default value

SPK\_GAIN & LINE\_GAIN : Range from -20 to 20

0000	00001	0010	0011	0100	0101	0110	0111
-20dB	-16 dB	-12 dB	-8 dB	-4 dB	0 dB	2 dB	4 dB
1000	1001	1010	1011	1100	1101	1110	1111
6 dB	8 dB	10 dB	12 dB	14 dB	16 dB	18 dB	20 dB

Loop Attenuation (CONF) : Range from 0 to 28 with 4dB/Step

000	001	010	011	100	101	110	111
0.dB	4 dB	8 dB	12 dB	16 dB	20 dB	24 dB	28 dB



**MX93032**

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## **MACRONIX INTERNATIONAL Co., LTD.**

**HEADQUARTERS:**

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