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Advanced embedded chatter box for physically challenging persons

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ABSTRACT

This paper details the proposal of a speaking device for physically challenging persons. This device will be a solution for people who are challenged with speaking and/or Hearing. Its purpose is to aid communication for the physically challenged. The Commercial version will be a compact size of 6.7" x 5.6 x 1.25". This will include basic 16 x 2 characters LCD – black on green 3.3V. Nonspeaking Individuals with problems with their fine motor skills can also use text-to speech to aid their abilities which include people who suffer from:

. ALS (Lou Gehrig's disease
· Traumatic Brain injury
· Laryngectomy
· First stroke patients. Also

Keywords - **APR9600IC, Zneo powers, Gpio's, timers**

I. INTRODUCTION

After seeing many deaf and dumb people around me a thought that can't we help physically challenging persons(Deaf, Dumb) by any simple device by using our technology with low cost and easy usage to give a alternative way to talk and hear for them brought me to do this paper. This paper details how to build a device which can speak out the text given to it by a speaker and helpful for dumb people as a mouth to them and its LCD display acts as a ear to the deaf people by making them to understand what the other person saying them. here in section 2 we discussed of some of the present solutions which are helpful to them, in section 3 we discuss of Possible implementation way, in section 4 we discuss of Apr9600ic which plays a major key role we even call it as heart of the device, a thing more of this ic is for future scope of this device we just need to change this ic with special features of inbuilt data with pure pronousation of the word where as this APR9600ic doesn't have those features, in section 5 we discuss of Zneo power Architecture.

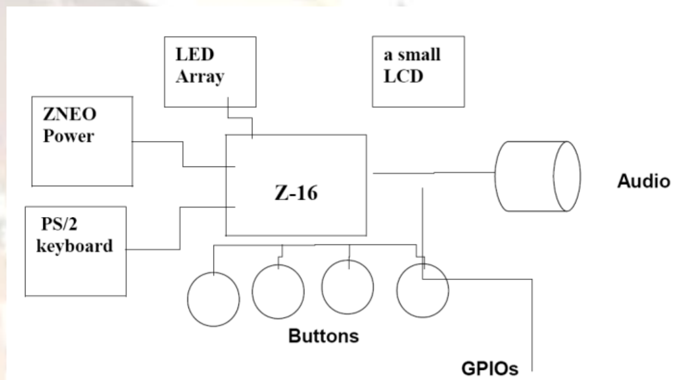


Fig1.Block diagram

II. PRESENT SOLUTION

2.1. Mobile ASL phones

The tool, which can be integrated to any high-end mobile phone with a video camera, is undergoing field tests involving 11 participants and the researchers plan to launch a larger field study this winter. "This is the first study of how deaf people in the US use mobile video phones," According to the Study, the engineers are now working to optimize compressed video signals for sign language, increasing the quality of the images around the face and hands to reduce the data rate to 30 kilobytes per second. To minimize the amount of battery power, the MobileASL phones employ motion sensors to determine whether sign language is being used, it said. Transmitting sign language as efficiently as possible increases affordability improves reliability on slower networks and extends battery life, even on devices that might have the capacity to deliver higher quality video. And the field test is allowing the team to see how people use the tool in their daily lives and what obstacles they encounter. Texting or email is currently the preferred method for distance communication of deaf and hearing-impaired people.

But the participants' experiences with the MobileASL phone are, in general, positive. The MobileASL system, the research says that, it could be integrated with the iPhone 4, the HTC Evo, or any device that has a video camera on the same side as the screen. The reason is that I bring it up again is because many people are still misleading to educate other people who want to learn the truth from the positive reinforcement I like to share with you again. I am getting fed up with these negative terms of deaf and dumb or Deaf and Mute over and over again.
Nowadays they are still seeing a very negative reinforcement about us being deaf and our speech reading or lip-readings that have in a mute, we feel more comfortable and are capable to express our true inner soul without being force us to speak with our Deaf voices with ASL. This does not success in our Deaf lives by their own audits attitudes. So therefore what do you want from us besides our Deaf voices/oral speaking that doesn't go anywhere. After all we did our part but that has not solved any issues. Since here research is going to say many people have no common sense since Deaf to deaf, deaf to deaf, deaf to latened Deaf and latened Deaf to Hearing do not understand each other because it's too much complicated for having to mislead and misunderstand without any ASL, and it also affect our Deaf education all along, as usual. We do read their lip movements. This exists with or without voices; however it doesn't take us anywhere. We all do not hear nor are we capable to hear everything.

This is with any kind of spoken languages with or without devices.
We must emphasize that "Mute" doesn't mean that we cannot speak without a voice. Because many of us do watch facial while we use our American Sign Language. Also, we do yell, scream, and make some noise that comes out with mumbling voices. It's not a mute at all so what is your point for labeling us MUTE which itself is as a very negative word? Also I prefer to use a mute (turn off my deaf voice) because it helps me focus, express, think, capable to communicate with the concept of ASL, to built up creative thinking, and get more ASL education to have it be proper be part of a learning process with our American Sign Language.

Matter of fact, that 'mute' is to turn off our Deaf voices; we prefer to use to avoid the confusion by inferring the true expression and feelings from oral speaking. So therefore, also, how is it determined that the kid needs a device anyway? D/deaf have done well without a device and there is so much evidence that Deaf people with a very healthy mental/physical were successful all along.

That is what ticks me off after all they screwed me and many deafies up since we were a little innocent children until today's it s still happens to Deaf children out there. I can speak fairly well but it doesn't take me anywhere so what the heck are you trying to do these Deaf children's lives for your own selfish/money reasons? Don't we realize that these devices and the need for it are manufactured like a robot being hearing? That is not normal/natural for us to hear the unnatural sounds and our being deaf as is. Without a device, some things that I am capable to hear naturally like a loud beep button or loud music or anything that it makes a real loud noise. I couldn't stand to hear the sounds anymore that are too much annoying to hear it. We can feel this most of the time. What irritates us is that people believe what the MD's (doctors) tell the parents without even thinking about it seriously. That's one of my biggest objections, there is NO guarantee, AND the person has to have their hearing that may exist taken away to make them almost 100% deaf in both sides. To take something away so it can be replaced just is freaking stupid or selfish by Audits people who don't care about us as being deaf and our mental health.

III. Implementation Plan

This describes how I will build my project; and what steps I will need to follow.

- Use small Z16 kit, get one from system admin.
- Acquire all components (LED, buttons, ZNEO Power, APR9600 text-to-speech IC, small LCD, and internal memory).
- Test APR9600 text-to-speech IC if needed.
- Building ZNEO power, and test life time if it continues after more than 1 hour.
- Check Display LED to confirm it is okay.
- Test Device driver (audio).
- Use 4 buttons.
- Text display on the APR9600 text-to-speech if it is okay.
- Draw simple milestone chart.
- Record all the 8alphabets in each ic which can be replayed when text is given.
- LCD for display of the text given.

IV. HARDWARE DESIGN

4.1. APR9600IC

4.1.1 Features:

• Single-chip, high-quality voice recording & playback solution
- No external ICs required
- Minimum external components

• Non-volatile Flash memory technology
- No battery backup required

• User-Selectable messaging options
- Random access of multiple fixed-duration messages
- Sequential access of multiple variable-duration messages

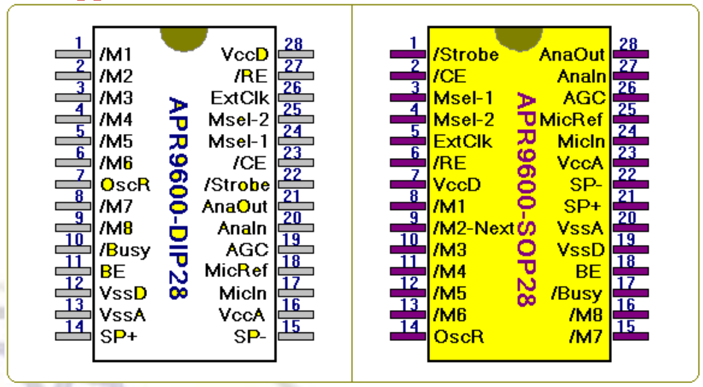
• User-friendly, easy-to-use operation- Programming & development systems not required
- Level-activated recording & edge-activated play back switches

• Low power consumption
- Operating current: 25 mA typical
- Standby current: 1 uA typical
- Automatic power-down

• Chip Enable pin for simple message expansion

4.1.2. General Description:

The APR9600 device offers true single-chip voice recording, non-volatile storage, and playback capability for 40 to 60 seconds. The device supports both random and sequential access of multiple messages. Sample rates are user-selectable, allowing designers to customize their design for unique quality and storage time needs. Integrated output amplifier, microphone amplifier, and AGC circuits greatly simplify system design. The device is ideal for use in portable voice recorders, toys, and many other consumer and industrial applications.
APLUS integrated achieves these high levels of storage capability by using its proprietary analog/multilevel storage technology implemented in an advanced Flash non-volatile memory process, where each memory cell can store 256 voltage levels. This technology enables the APR9600 device to reproduce voice signals in their natural form. It eliminates the need for encoding and compression, which often introduce distortion.



PS : The APR9600 DIP & SOP is not [PIN TO PIN]

4.1.3 Functional Description:

APR9600 block diagram is included in order to describe the device's internal architecture. At the left hand side of the diagram are the analog inputs. A differential microphone amplifier, including integrated AGC, is included on-chip for applications requiring use. The amplified microphone signals fed into the device by connecting the ANA_OUT pin to the ANA_IN pin through an external DC blocking capacitor. Recording can be fed directly into the ANA_IN pin through a DC blocking capacitor, however, the connection between ANA_IN and ANA_OUT is still required for playback. The next block encountered by the input signal is the internal anti-aliasing filter. The filter automatically adjusts its response according to the sampling frequency Selected so Shannon's Sampling Theorem is satisfied. After anti-aliasing filtering is accomplished the signal is ready to be clocked into the memory array. This storage is accomplished through a combination of the Sample and Hold circuit and the Analog Write/Read circuit. These circuits are clocked by either the Internal Oscillator or an external clock source. When playback is desired the previously stored recording is retrieved from memory, low pass filtered, and amplified as shown on the right hand side of the diagram. The signal can be heard by connecting a speaker to the SP+ and SP- pins. Chip-wide management is accomplished through the device control block shown in the upper right hand corner. Message management is provided through the message control block represented in the lower center of the block diagram.
More detail on actual device application can be found in the Sample Application section. More detail on sampling control can be found in the Sample Rate and Voice Quality section. More detail on Message

management and device control can be found in the Message Management section.

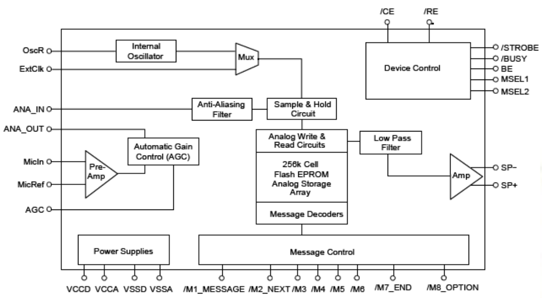


Figure 2 APR9600 Block Diagram

4.2 Message Management:

4.2.1 Message Management General Description

Playback and record operations are managed by on-chip circuitry. There are several available messaging modes depending upon desired operation. These message modes determine message management style, message length, and external parts count. Therefore, the designer must select the appropriate operating mode before beginning the design. Operating modes do not affect voice quality; for information on factors affecting quality refer to the Sampling Rate & Voice Quality section. The device supports five message management modes (defined by the MSEL1, MSEL2 and /M8_OPTION pins shown in Figures 1 and 2). Random access mode with 2, 4, or 8 fixed-duration messages Tape mode, with multiple variable-duration messages, provides two options:

- Auto rewind
- Normal

Modes cannot be mixed. Switching of modes after the device has recorded an initial message is not recommended. If modes are switched after an initial recording has been made some unpredictable message fragments from the previous mode may remain present, and be audible on playback, in the new mode. These fragments will disappear after a Record operation in the newly selected mode. Table 1 defines the decoding necessary to choose the desired mode. An important feature of the APR9600 Message management capabilities is the ability to audibly prompt the user to change in the device's status through the use of "beeps" superimposed on the device's output. This feature is enabled by asserting a logic high level on the BE pin.

Table 1

Mode	MSEL1	MSEL2	/M8_OPTION
Random Access 2 fixed duration messages	0	1	Pull this pin to VCC through I00K resistor
Random Access 4 fixed duration messages	1	0	Pull this pin to VCC through I00K resistor
Random Access 8 fixed duration messages	1	1	The /M8 message trigger becomes input pin
Tape mode, Auto rewind operation	0	0	0
Tape mode, Normal operation	0	0	1

4.2.2 Random Access Mode

Random access mode supports 2, 4, or 8 Message segments of fixed duration. As suggested recording or playback can be made randomly in any of the selected messages. The length of each message segment is the total recording length available (as defined by the selected sampling rate) divided by the total number of segments enabled (as decoded in Table1). Random access mode provides easy indexing to message segments.

4.2.2A Functional Description of Recording in Random Access Mode

On power up, the device is ready to record or playback in any of the enabled message segments. To record,/CE must be set low to enable the device and /RE must be set low to enable recording. You initiate recording by applying a low level on the message trigger pin that represents the message segment you intend to use. The message trigger pins are labeled /M1_MESSAGE - /M8_OPTION on pins 1-9 (excluding pin 7) for message segments 1-8 respectively. Note: Message trigger pins of M1_MESSAGE,/M2_NEXT, /M7_END, and /M8_OPTION, have expanded names to represent the different functionality that these pins assume in the other modes. In random access mode these pins should be considered purely message trigger pins with the same functionality as /M3, /M4, /M5, and /M6. For a more thorough explanation of the functionality of device pins in different modes please refer to the pin description table that appears later in this document. When actual recording begins the device responds with a single beep (if the BE pin is high to enable the beep tone) at the speaker outputs to indicate that it has started recording. Recording continues as long as the message pin stays low. The rising edge of the same message trigger pin during record stops the recording operation (indicated with a single Beep).If the message trigger pin is held low beyond the end of the maximum allocated duration, recording stops automatically (indicated with two beeps), regardless of the state of the message trigger pin. The chip then enters low-power mode until the message trigger pin returns high. After the message trigger pin returns to high, the chip enters standby mode. Any subsequent high to low transition on the same message trigger pin will initiate recording from the beginning of the same message segment. The entire previous message is then overwritten by the new message, regardless of the duration of the new message. Transitions on any other message trigger pin or the /RE pin during the record operation are ignored until after the device enters standby mode.

4.1.2B Functional Description of Playback Random Access Mode

On power up, the device is ready to record or playback, in any of the enabled message segments. To playback,/CE must be set low to enable the device and /RE must be set high to disable recording & enable playback. You initiate playback by applying a high to low edge on the message trigger pin that represents the message segment you intend to playback. Playback will continue until the end of the message is reached. If a high to low edge occurs on the same message trigger pin during playback, playback of the current message stops immediately. If a different message trigger pin pulses during playback, playback of the current message stops immediately (indicated by one beep) and playback of the new message segment begins. A delay equal to 8,400 cycles of the sample clock will be encountered before the device starts playing the new message. If a message trigger pin is held low, the selected message is played back repeatedly as long as the trigger pin stays low. A period of silence, of duration equal to 8,400 cycles of the sampling clock, will be inserted during looping as an indicator to the user of the transition between the end and the beginning of the message.

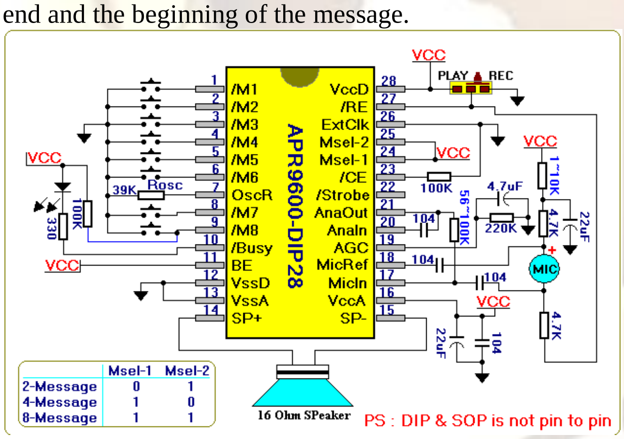


Figure 3 Random Access Mode : 2 / 4 / 8 Message

4.3 Signal Storage:

The APR9600 samples incoming voice signals and stores the instantaneous voltage samples in non-volatile FLASH memory cells. Each memory cell can support voltage ranges from 0 to 256 levels. These 256 discrete voltage levels are the equivalent of 8-bit (28=256) binary encoded values. During playback the stored signals are retrieved from memory, smoothed to form a continuous signal, and then amplified before being fed to an external speaker.

4.4 Sampling Rate & Voice Quality:

According to Shannon's sampling theorem, the highest possible frequency component introduced to the input of a sampling system must be equal to or less than half the sampling frequency if aliasing errors are to be eliminated. The APR9600 automatically filters its input, based on the selected sampling frequency, to meet this requirement. Higher sampling duration capabilities of the device, but they also reduce incoming signal bandwidth. The APR9600 accommodates sampling rates as high as 8 kHz and as low as 4 kHz. You can control the quality/duration trade off by controlling the sampling frequency. An internal oscillator provides the APR9600 sampling clock. Oscillator frequency can be Changed by changing the resistance from the OscR pin to GND. Table 2 summarizes resistance values and the corresponding sampling frequencies, as well as the resulting input bandwidth and duration. Rates increase the bandwidth and hence the voice quality, but they also use more memory cells for the same length of recording time. Lower sampling rates use fewer memory cells and effectively increase the

Table 2 Resistance Values & Sampling Frequencies

Resistance	Sampling Frequency	Input Bandwidth	Duration
84 K	4.2 kHz	2.1 kHz	60 sec
38 K	6.4 kHz	3.2 kHz	40 sec
24 K	8.0 kHz	4.0 kHz	32 sec

4.5 Automatic Gain Control (AGC):

The APR9600 device has an integrated AGC. The AGC affects the microphone input but does not affect the ANA_IN input. The AGC circuit insures that the input signal is properly amplified. The AGC works by applying maximum gain to small input signals and minimum gain to large input signals. This assures that inputs of varying amplitude are recorded at the optimum signal level. The AGC amplifier is designed to have a fast attack time and a slow decay time. This timing is controlled by the RC network connected to pin 19. A value of 220K and 4.7uF has been found to work well for the English language. Be aware that different languages, speakers from different countries, and music may all require modification of the recommended values for the AGC RC network.

4.6 Sampling Application :

The following reference schematics are included as examples of how a recording system might be designed. Each reference schematic shows the device incorporated in one of its three main modes: Random Access, Tape mode – Normal option, and Tape mode – Auto Rewind option. Note that in several of the applications either one or all of the /BUSY, /STROBE,or /M7_END pins are connected to LEDs as indicators of device status. This is possible because all of these pins and signals were designed to have timing compatible with both microprocessor interface and manual LED indication. A bias must be applied to the electret microphone in order to power its built-in circuitry. The ground return of this bias network is connected to the /Busy. This configuration saves power when record mode.Both pins 18 and 19, MicIn and MicRef, must be AC coupled to the microphone network in order to block the DC biasing voltage. Figure 3 shows the device configured in random access mode. The device is using eight Message segments, the maximum available, in this mode. Note that message trigger pins that are not used, for modes with less than eight segments, can be left unconnected with the exception of pin /M8_OPTION which should be pulled to VCC through a 100k resistor.

4.7 Electrical Characteristics:

The following tables list absolute maximum ratings, DC Characteristics, and Analog Characteristics for the APR9600 device.

Table 1 Absolute Maximum Rating

Item	Symbol	Condition	Min	Max	Unit
Power Supply voltage	VCC	TA = 25℃	-0.3	7.0	V
Input Voltage	VIN	IIN<20mA	-1.0	Vcc + 1.0	V
Storage Temperature	TSTG	-	-65	150	℃
Temperature Under Bias	TBS	-	-65	125	℃
Lead Temperature	TLD	<10s	-0.3	300	℃

Table 2 DC Characteristics

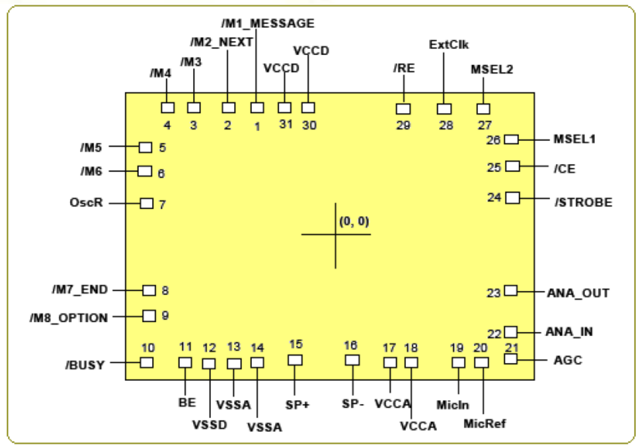
Item	Symbol	Condition	Min	Typ	Max	Unit
Power Supply voltage	VCC	TA = 25℃	4.5	-	6.5	V
Input High Voltage	VIN	-	2.0	-		V
Input Low Voltage	VIL	-	-	-	0.8	V
Output High Voltage	VOH	IOH=-1.6mA	2.4	-	-	V
Output Low Voltage	VOL	IOL=+4.0mA	-	-	0.45	V
Input Leakage Current	VIH=Vcc	-	-	-	1.0	uA
Input Leakage Current	IIL	VIL=Vss	-1.0			uA
Output Tri-state Leakage Current	IOZ	VOUT=Vcc or VOUT=VSS	-1.0		1.0	uA
Operating Current Consumption	ICC	Internal Clock No Load	-	25	-	mA
Standby Current Consumption	ICCS	No Load	-	1.0	-	uA

Table 3 Analog Characteristics

Item	Symbol	Condition	Min	Typ	Max	Unit
MicIn Input Voltage	VMI				30	mVP-P
MicIn Input Resistance	RMI			15		kΩ
MicIn Amp Gain (1)	GMI1	AGC=2.25V		30		dB
MicIn Amp Gain (2)	GM2	AGC=3.8V		-2		dB
ANA_IN Input Voltage	VANI				140	mVP-P
ANA_IN Input Resistance	RANI			500		kΩ
ANA_IN Amp Gain	GANI	ANA_IN to SP+/-		10		dB
AGC Output Resistance	RAGC			225		KΩ
Sp+/- Output Power	PSP	RSP+/-=16Ω		12.2		MW
Voltage Amplitude across SP+/-	VSP	RSP+/-≧16Ω		1.4		VP-P

4.8. Bonding Pad Diagram and Bonding Pad Coordinates :

Figure 6 APR9600 Die Bonding Pad Diagram



Die Dimensions X-Axis: 212 +/- 1 mils (X-Axis: 5450 um)
Y-Axis: 176 +/- 1 mils (Y-Axis: 4550 um)
Die Thickness 13.8 +/- 1.0 mils (350 +/- 25 um)
Pad Opening 4.3 mils (110 um)

V. ZNEO POWER

ZNEO Z16F Series MCU is a powerful 16-bit CISC microcontroller that outperforms most RISC microcontrollers in its class. The ZNEO Z16F Series boasts a unique architecture that provides the power, punch, and performance of a 32-bit, with the code, current efficiency, and cost of a 16-bit. The ZNEO CPU boasts a highly optimized instruction set that achieves higher performance per clock cycle, with less code space and lower overhead. All ZNEO Z16F Series products are RoHS-compliant.

5.1.Z16F2810 Features

- 20MHz ZNEO Single-Cycle CISC microprocessor core

- 128KB internal Flash program memory with 16-bit access and in-circuit programming capability
- 4KB internal RAM with 16-bit access
- 12-channel, 10-bit analog-to-digital converter (ADC)
- Operational Amplifier
- Analog Comparator
- 4-channel DMA controller supports internal or external DMA requests
- Two full-duplex 9-bit UARTS with support for LIN and IrDA
- 5.5MHz Internal Precision Oscillator
- I2C master-slave controller
- Enhanced Serial Peripheral Interface (ESPI) controller.
- 12-bit PWM module with three complementary pairs or six independent PWM outputs with dead-band generation and fault trip input

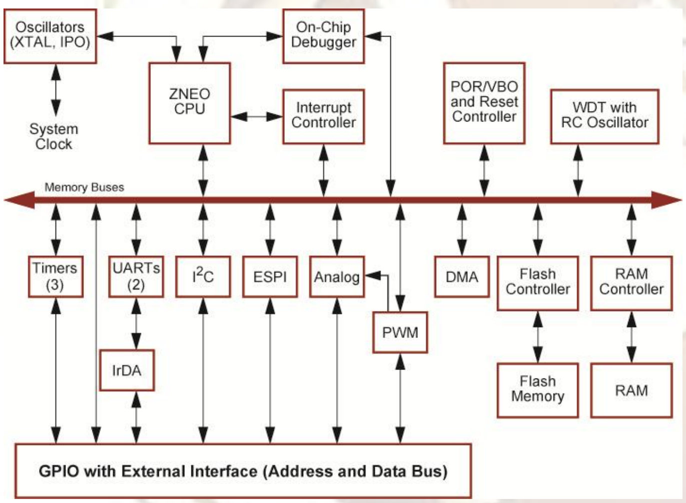


Fig2. ZNEO POWER ARCHITECTURE

VI.CONCLUSIONS&FURTHERRECOMMENDATIONS

Thus literature survey to advanced embedded chatter box for physically challenging persons has been done and their data sheets corresponding to the chatter box, i.e mainly controller, A/D converters and related software keil is been verified with a demo program with interfacing modules as leds,and manly speak jet ic which acts like main interfacing component for the physically challenged to text to speech converter, hardware implementation has to be done for verifying its functional properties. Finally it may be one of the useful devices for deaf people to understand what the others are saying by LCD display and dumb people can speak with the help of this device. In future an Ic which may be inbuilt with all speaking futures can be used instead of APR9600 ic and may be more useful for physically challenging persons.

ACKNOWLEDGEMENTS

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SHORT BIOGRAPHY

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