VOICE RECORD/PLAYBACK MODULE

by ROBERT PENFOLD

Your very own personal "voice box". Will digitally record up to 16 seconds of sound.

The days when gadgets capable of understanding what we say or able talk back to us were in the realms of science fiction are now well and truly behind us. This article is being dictated into a PC using (more or less) normal speech, and when checking the final piece for errors the computer will be instructed to read the text back.

It is not only in the computer world that this sort of voice interface is starting to emerge. All manner of electronic gadgets that can understand simple commands and (or) talk back to us are becoming available.

The device featured here is a general-purpose voice recording and playback module that can handle up to 16 seconds of speech. It can be used as a complete project in its own right, and it then operates as a simple messaging system.

In order to record a message you simply press a button, say a short message such as *"taken the cat to the vet, back at about 4-30"* and then release the button. Your message can then be played back as many times as required simply by pressing a second pushbutton switch. New messages can be recorded over existing ones as and when required.

RED ALERT

Another way of using the unit is as an alternative to an LED (light-emitting diode) indicator or a low-power audio alarm generator circuit. When used in this way you must first record a suitable message into the unit, such as a *"warning – maximum temperature exceeded"* or *"red alert – this is not a drill"* if your sense of humor gets the better of you!

A big advantage of this system is that you can use any words you like, and you are not restricted to very brief messages. When the module is activated the message can either be played back just once, or it can be repeated for as long as power is applied to the module, as preferred.

The basic recording and playback circuit is designed to operate from a supply potential of about 5V, but an optional voltage regulator enables the circuit to operate over a supply voltage range of about 7V to 15V. The module has a built-in electret microphone insert and will directly drive a small loudspeaker having an impedance

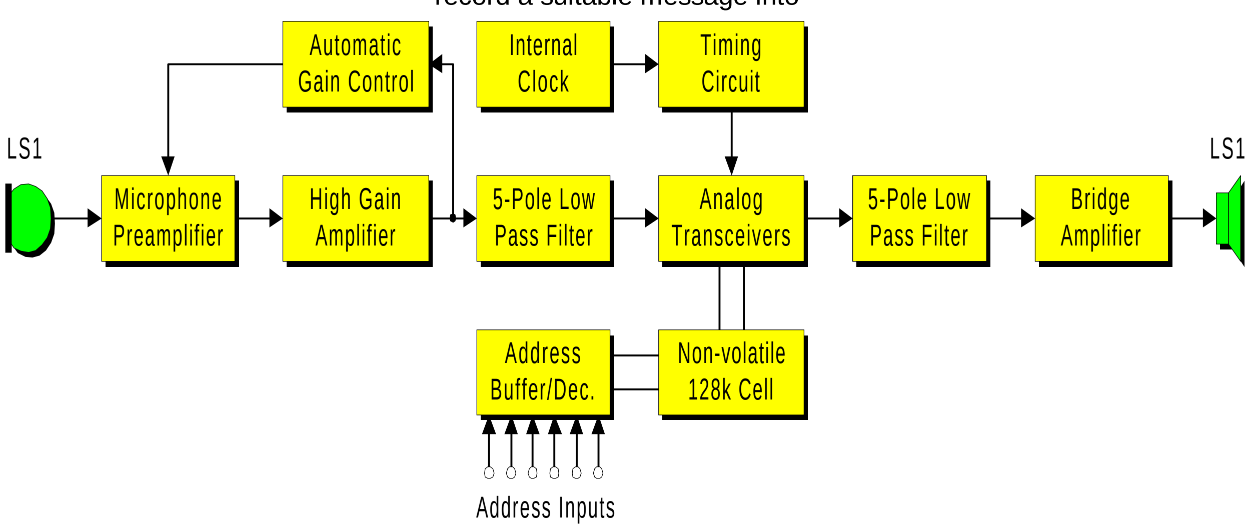


Fig.1. Simplified block schematic diagram for the ISD1416 voice chip.

rating of 16 ohms or more.

Current consumption of the module is insignificant when it is in the standby mode, but the non-volatile memory retains its contents even when the module is switched off. This avoids the need for any form of battery back-up circuit.

A unit of this type could easily be very complex and large, but by using a dedicated integrated circuit the component count of this circuit is kept to a minimum. The speech quality of many “talking” circuits leaves a lot to be desired, but in this case the quality is very good due to the use of a recorded voice rather than speech synthesis techniques. In fact the quality is surprisingly good, and is limited mainly by the quality of the microphone and loudspeaker used rather than the recording and playback circuit.

Although this module is quite simple, in most cases it will require a certain amount of technical knowledge in order to use it properly. Consequently, it cannot really be regarded as a beginner's project.

SYSTEM OPERATION

This project is based on the ISD1416 “ChipCorder” integrated circuit from ISD. It is an extremely complex chip that provides all the active circuitry needed for this application. The internal arrangement of this chip is shown in the simplified block diagram form of Fig 1.

The output level from an electret microphone is extremely small, and a large amount of amplification is needed in order to bring the signal to a level that will drive the analogue-to-digital converter properly. A low noise preamplifier followed by a high gain amplifier provides this amplification.

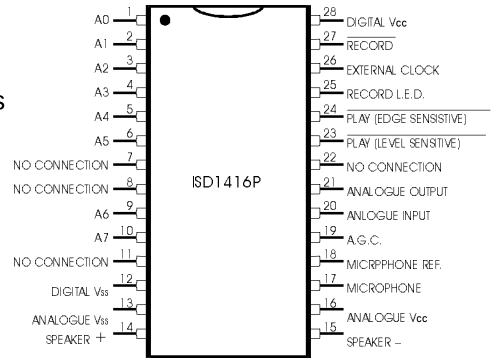


Fig.2. Pinout details for the ISD1416 speech recorder chip.

It is likely that the input level will vary enormously depending on factors such as the particular microphone used, the distance from the user to the microphone, and the loudness of the user's voice. The input stages of the device therefore incorporate a simple but effective AGC (automatic gain control) action that prevents overloading and the extreme distortion this would produce.

ACTIVE FILTER

The next stage in the main signal path is a 5-pole lowpass filter. In common with other digital recordings systems, this one uses a system of sampling the input signal at regular intervals.

The sampled values are stored in memory, and then played back through a digital-to-analogue converter during playback. This converts the stored values back to the original sample voltages, and recreates the original signal.

One slight problem with any sampling system is that any input signals close to the sampling frequency produce a very severe form of distortion known as ”aliasing” distortion. In this case, the sampling frequency is just eight kilohertz (8kHz), which is well within the audio range. This makes it important to have a very effective filter to attenuate input signals at more than about one-third to one-half of the sampling frequency.

This is the purpose of the 5-pole active lowpass filter at the input of the analogue-to-digital converter. With a sampling frequency of 8kHz, the maximum signal frequency that can be handled by the system is only about 3kHz or 4kHz, but this is perfectly adequate for good results with speech signals.

The ISD1416 data sheet is very vague about the analog-to-digital and digital-to-analog converters, which are simply referred to as “analog transceivers”, but the audio quality of the device would suggest that these have a resolution of at least 8-bits. The EEPROM has a capacity of 128 kilobytes, which together with the sampling rate of 8kHz gives a maximum message duration of 16 seconds.

Typically, the memory retains its contents for 100 years and has a lifetime of one hundred thousand record cycles. Some of the address inputs are accessible, but in normal operation it is not necessary to take direct control of the memory. The internal control circuits automatically start playback at the beginning of the sample and halt it at the end.

It is not necessary to use the full 16 seconds of message time, and messages can have any duration up to the 16-second maximum available. The device has internal clock oscillator and timing circuits that provide suitable control signals to the converters, memory, etc.

OUTPUT

On the output side of the converter stage there is another 5-

pole lowpass filter and a small audio power amplifier. Any sampling method of recording inevitably results in a stepped output waveform as the signal jumps from one sample level to the next.

This effectively provides an output signal that is modulated with the sampling frequency. In this case the sampling frequency is well within the audio range, and could produce a clearly audible tone on the output. The lowpass filter at the output smoothes the signal to remove the stepping, and in doing so it also removes any audible breakthrough of the clock signal.

The power amplifier at the output of the device is a bridge circuit that can directly drive a loudspeaker that has an impedance of 16 ohms or more. Using a bridge circuit enables a reasonable output power to be obtained despite the fact that the supply potential is only 5V and the loudspeaker is a high impedance type.

The ISD1416 also includes a substantial amount of control logic that enables the recording and playback functions to be controlled by just two pushbutton switches. These control circuits also govern such things as whether the device operates in the one-shot mode or plays back samples continuously.

PINOUTS

Pinout details for the ISD1416, which is contained in a standard 28-pin DIL encapsulation, is shown in Fig.2. There are separate supply pins for the analogue and digital circuits, but in normal use these are fed from a common supply.

Pins designated A0 to A7 are the address inputs, but in many applications these can simply be connected to the 0V supply rail and otherwise ignored. If the two most significant bits (A6 and A7) are high, the address inputs control the operating mode of the device.

Address input A3 is one of the most useful, and controls whether the chip operates in single-shot mode or loops continuously. This input is taken high in order to set the device into the continuous loop mode.

There is provision for an external clock circuit, and the clock signal can be applied to pin 26. This facility is not normally required, and pin 26 is then connected to the 0V supply rail.

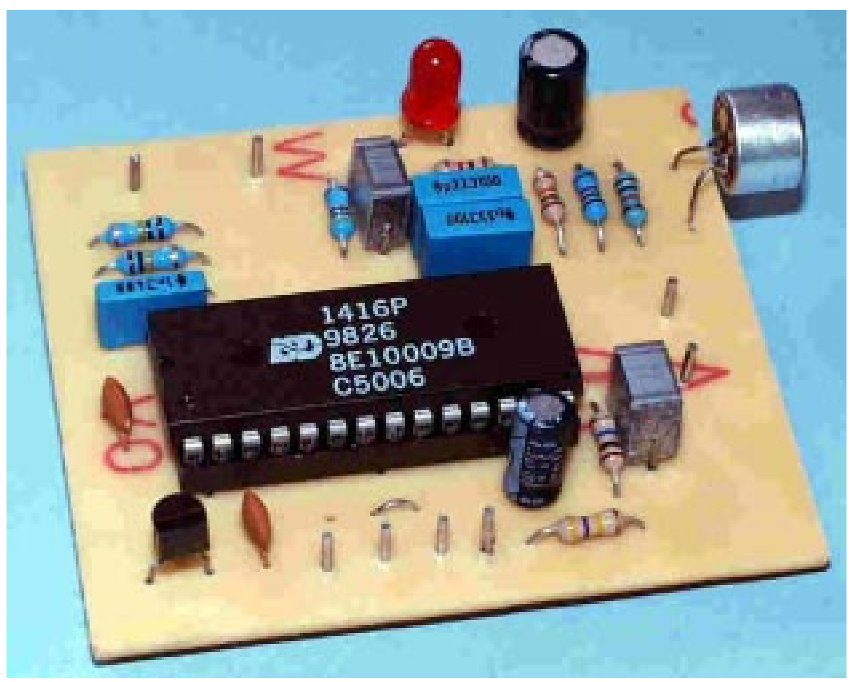
RECORD/PLAYBACK

Recording and playback are controlled via three inputs, one of which is used to produce record cycles (pin 27). Like the other control inputs, this pin is normally held high. However, it must be taken low and kept low while the message is recorded.

An open collector output at pin 25 is switched on during this period, and can be used to operate an LED indicator which confirms that the recording cycle is proceeding normally. If the recording has not been completed by the end of the 16-second maximum recording period, the LED will switch off to indicate that the recording has finished.

Pin 23 and pin 24 control playback, and in most applications it is the edge sensitive input at pin 24 that is utilized. Taking this input low, even momentarily, results in the complete message being played back. When using the level sensitive input at pin 23 the message is only played back while the input is held low, and the message will be truncated if this input is returned to the high state prematurely.

On the audio side of things there are differential inputs at pin 17 and pin 18. Differential inputs can help to ease problems with stray pick up of noise, but in most applications there will be no long microphone cables and this will be purely academic.



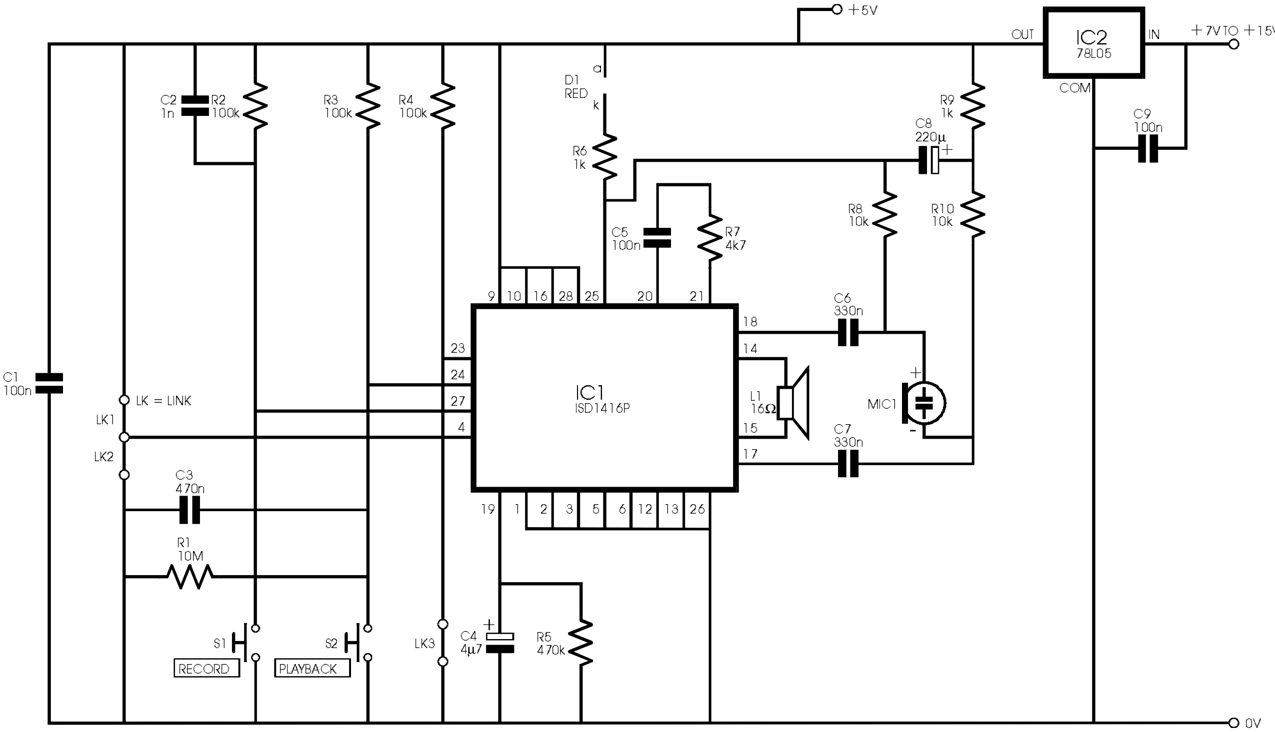


Fig.3. Complete circuit diagram for the Voice Record/Playback Module.

A resistor and capacitor network connected to pin 19 controls the decay time of the AGC circuit. The output of the preamplifier and the input of the amplifier stage are available at pins 20 and 21 respectively, and an external capacitive coupling is required here. Four pins of the device have no internal connections.

CIRCUIT OPERATION

The full circuit diagram for the Voice Record/Playback Module appears in Fig.3. In practical applications not all of the components and links shown in the circuit diagram will be required. We will deal first with the components that MUST always be included.

The microphone MIC1 is capacitively coupled to the inputs of the voice chip IC1 by way of capacitors C6 and C7. Unlike most types of microphone, the electret variety has a built in preamplifier that requires a power source.

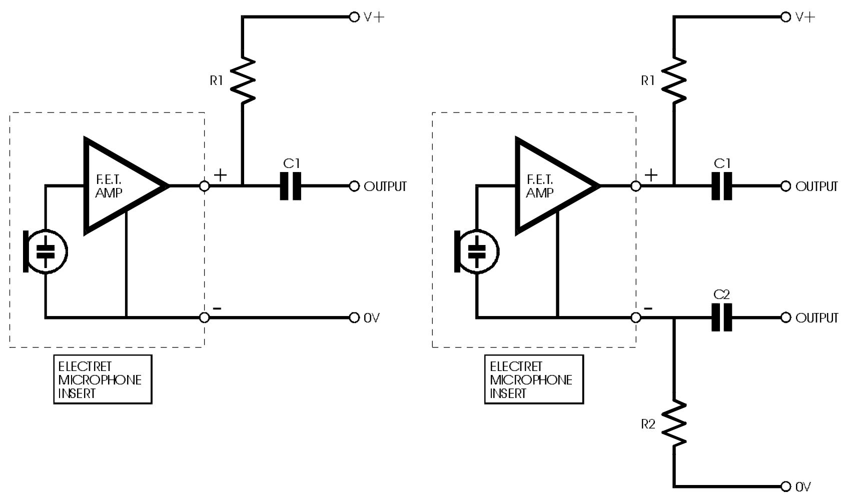


Fig.4. (left) Normal method of using an electret microphone insert, and (right) producing differential output signals.

Modern electret microphones usually have just two terminals, and require an external load resistor for the preamplifier stage, as shown in of Fig.4a. The preamplifier is usually a simple JFET (junction field-effect transistor) circuit that will operate from a low supply potential and draws little supply current. In most cases the circuit will operate at supply potentials as low as one volt or so, and with a supply current of less than 100 microamps.

INPUT CIRCUIT

The input circuit used here may look a little unusual, but it uses the method recommended in the data sheet for the ISD1416. This produces differential output signals using the method of connection shown in Fig.4b.

Resistor R9 and capacitor C8 form a decoupling network in the supply to the microphone circuit, see Fig.3. This avoids problems with feedback through the supply lines, and digital noise being coupled into the audio path via the supply.

The 0V supply for the microphone circuit is obtained from pin 25 of IC1, which results in the microphone circuit being switched off except when a recording is being made. This maintains the very low standby supply current of less than a microamp.

Diode D1 is the Recording indicator LED and R6 is its current limiter resistor. Incidentally, this LED also flashes briefly at the end of each playback cycle.

Capacitor C5 and resistor R7 couple the output of the microphone preamplifier to the input of the amplifier stage. The *C-R* timing network for the AGC circuit is comprised of C4 and R5. It is necessary to use a fairly long time constant here in order to avoid rapid changes in gain and consequent distortion.

Because the output amplifier is a bridge circuit it is not necessary to use a coupling capacitor in series with loudspeaker LS1. Under standby conditions both outputs are about half the supply potential, giving 0V across the loudspeaker.

In operation the outputs provide anti-phase signals (i.e. as one output goes more positive the other goes negative by an identical amount). This gives a maximum output voltage that is twice as high as using a single-ended output stage, and in theory the peak-to-peak output voltage can be double the supply potential.

In terms of output power, a bridge circuit gives up to four times the output of an equivalent single-ended circuit. Although the circuit only operates from a 5V supply, using a high impedance (about 64 ohms) loudspeaker provides adequate volume for most purposes.

If the unit will be used in a noisy environment it would be better to use a 16 ohm impedance loudspeaker, but a component of this impedance is unlikely to be available. Using two 8 ohm impedance loudspeakers connected in *series* is probably the best option. ***Note that using a loudspeaker having an impedance of less than 16 ohms could damage IC1.***

SUPPLY NEEDS

The ISD1416 is designed to operate from a 5V supply, and voltage regulator IC2 plus capacitor C9 are unnecessary if a supply of about 4·5 to 5·5V is available. If the unit is to be battery powered, three AA cells in a holder provide a nominal 4·5V supply and seem to give good results. Due to the very low quiescent current consumption of no more than 10µA (and typically just 0·5µA) it is unnecessary to use an on/off switch if the unit is used in a stand-alone application.

If the module is used instead of an LED indicator and has to operate from a supply of about 7V to 15V, it *must* be powered via IC2, and both IC2 and C9 *must* be included. Note that the standby current consumption of the circuit will be up to a few milliamps if IC2 is included, due to the current consumption of IC2 itself. This factor should not be important when the module is used in place of an LED indicator, because it should be switched off for the majority of the time.

Link LK1 is included if the unit must operate in the mode where it loops continuously, repeating the message for as long as power is applied to the cir-

COMPONENTS

Resistors
R1 10M
R2, R3, R4 100k (3 off)
R5 470k
R6, R9 1k (2 off)
R7 4k7
R8, R10 10k (2 off)
All 0.25W 5% carbon film

Capacitors
C1, C9 100n disc ceramic (2 off)
C2 1n polyester
C3 470n polyester
C4 4u7 radial electrolytic, 50V
C5 100n polyester
C6, C7 330n polyester (2 off)
C8 200u radial electrolytic, 10V

Semiconductors
D1 red panel LED
IC1 ISD1416P voice record/playback
IC2 78L05 +5V 100mA voltage regulator

Miscellaneous
S1, S2 pushbutton switch, push-to-make, release-to-break (2 off)
MIC1 electret microphone
LS1 moving coil loudspeaker (16 ohms or more -- see text)

Printed circuit board available from the *EPE Online store*, code 7000225 (**www.epemag.com**); 28-pin DIL socket; multistrand connecting wire, solder pins, solder, etc.

Some components are not required depending on the mode of operation (see text for more details).

Approx. Cost Guidance Only **$40**

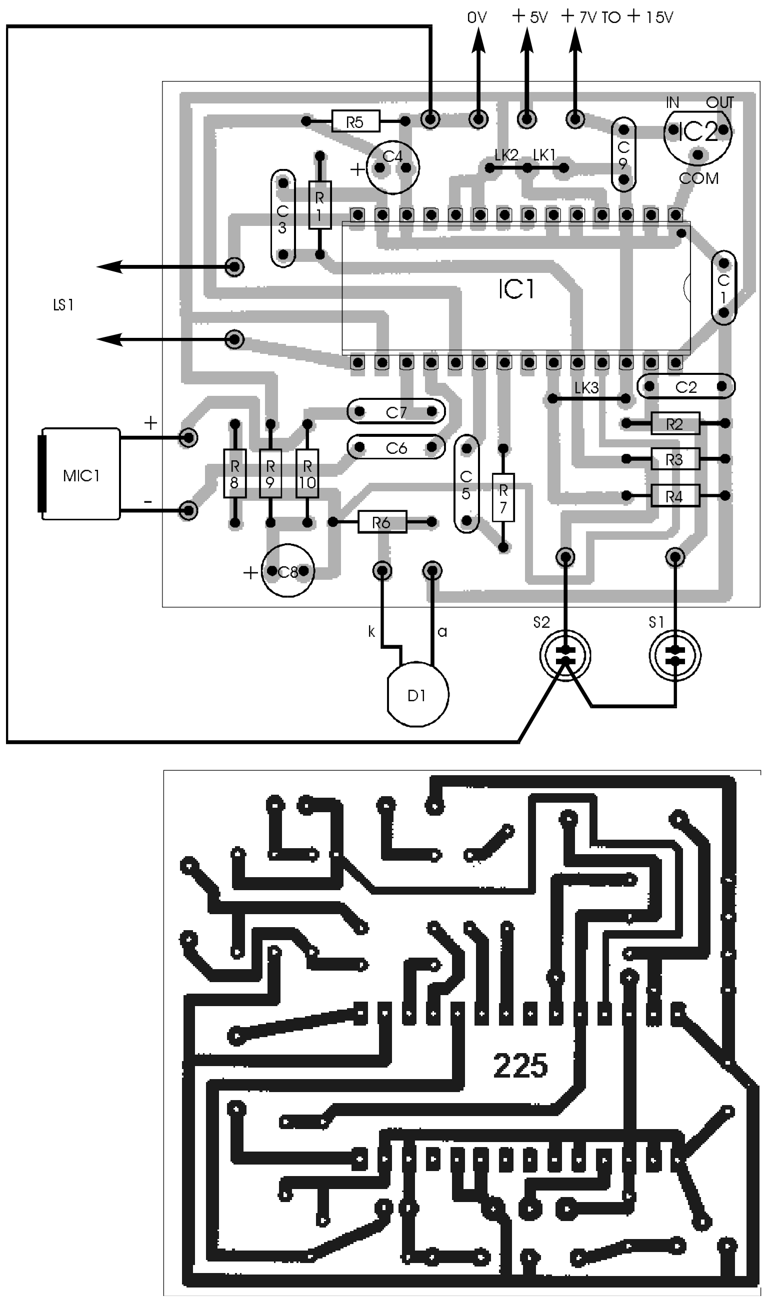


Fig.5. Voice Record/Playback Module printed circuit board component layout, wiring and (approximately) \full-size copper foil master pattern.

cuit. For single-shot operation include link LK2 instead. One or other of these link-wires must be included, **but obviously not both**.

If the module is used as a stand-alone device for handling messages, both switch S1 and S2 should be included. S1 is operated while a message is recorded, and S2 is pressed briefly in order to play back messages.

These switches could be included if the unit is used in place of an LED indicator, but they are not really needed. A crocodile clip lead or virtually any short piece of wire can be used to connect the appropriate two pins on the circuit board while your message is recorded. The same method can be used to trigger the unit to check that your message has been recorded properly.

LED INDICATOR

Where the unit is used in place of an LED indicator, it is clearly necessary to have the circuit trigger automatically when it is powered-up. One way of achieving this is to include capacitor C3 and resistor R1.

Capacitor C3 keeps pin 24 of IC1 low for several milliseconds after power-up, and this triggers it into a playback cycle. The message will, of course, be played back repeatedly if link LK1 is included. Resistor R1 discharges C3 when the power is removed so that the unit is soon ready to trigger again when power is restored.

Access to the level sensitive “play” input at pin 23 is provided, or link LK3 can be included so that this input is permanently held low. On the face of it this provides another means of automatically triggering the unit when it is powered-up, but this does not always seem to have the desired effect, and the author would recommend using R1 and C3 where automatic triggering is required.

CONSTRUCTION

The component layout and (approximately) actual size copper pattern for the printed circuit board (PCB) are shown in Fig.5. This PCB is available from the *EPE Online Store* (code 7000225) at **www.epemag.com**

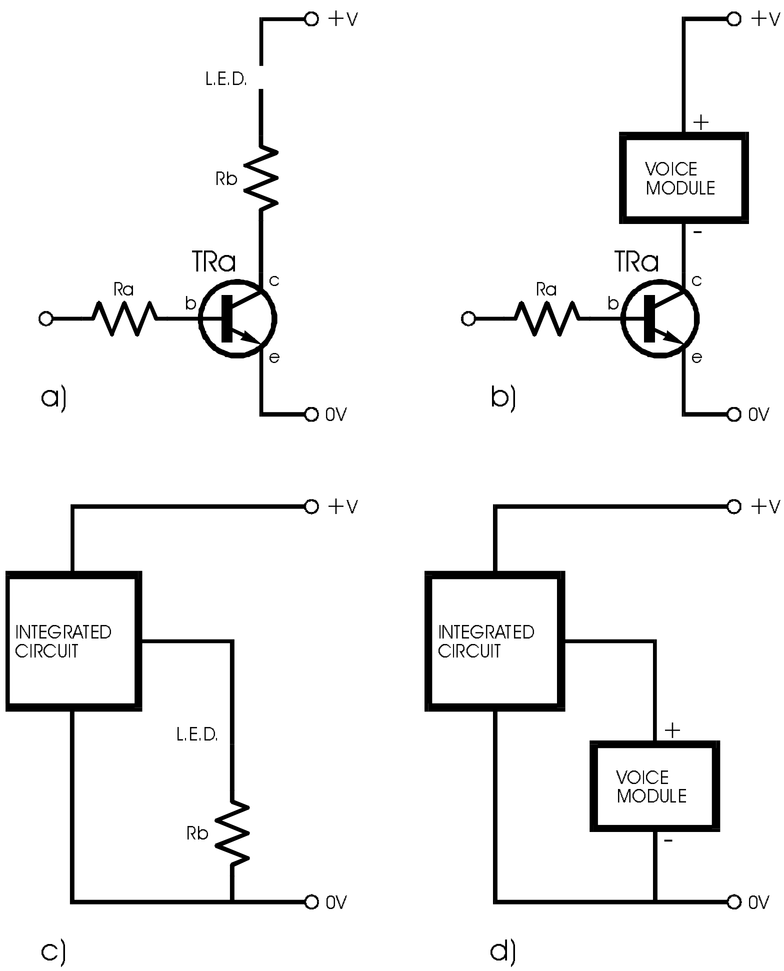


Fig.6. Two examples of using the module in place of an LED.

The board is fairly straightforward to construct, working from the smallest to largest component, but the ISD1416P is a CMOS device and is not particularly cheap. It is therefore essential to adhere to the standard anti-static handling precautions.

You *must* use a holder for the "voice" chip IC1, but *do not* plug it into its socket until the circuit board is otherwise complete. Try to touch the pins as little as possible, and keep the device away from any obvious sources of static charges such as computer monitors and television sets.

The electret microphone must be connected with the right polarity if it is to function well. The lead that connects to the metal case of the insert is usually the negative supply terminal, but where possible this point should be checked using the manufacturers or retailers literature.

OPERATING CHOICE

As pointed out previously, some of the components and link-wires will not be required, and which ones that have to be left out depends on your precise application. The following examples should help to clarify matters.

Stand-alone Messaging System

Omit IC2, C4, C9, link LK1, link LK3, and R1. Power the unit from a 4·5V battery and use the 0V and 5V supply inputs.

Indicator L.E.D. Replacement for 5V Supply

Omit IC2, C9, and link LK3. Include pushbutton switches S1 and S2 temporarily while the module is programmed and checked, or improvise with pieces of wire. For one-shot operation include link LK2 and omit link LK1. For continuous looping omit link LK2 and include link LK1. Use the 0V and 5V supply inputs.

Indicator L.E.D. Replacement for 7V to 15V Supply

Omit link LK3. Include switches S1 and S2 temporarily while the module is programmed and checked, or improvise with pieces of wire. For one-shot operation include link LK2 and omit link LK1. For continuous looping omit link LK2 and include link LK1. Use the 0V and 7V to 15V supply inputs.

CASING-UP

If the unit is constructed as a stand-alone messaging system, it will obviously have to be fitted in its own case and virtually any small to medium size case should accommodate everything. A grille is required for the loudspeaker, and there are various ways of producing this.

The standard approach is to make a large round cutout that is slightly smaller than the diameter of the loudspeaker. A piece of speaker cloth or fret is then glued in place behind the cutout. A simple alternative is to drill a matrix of holes about four or five millimeters in diameter, but this needs to be done very carefully if a neat appearance is to be obtained.

Miniature loudspeakers invariably lack any provision for screw fixing, leaving little alternative to gluing them in place. Only apply the adhesive to the front rim of the loudspeaker, taking care not to smear any over the diaphragm. Any good quality general-purpose adhesive should do the job quite well.

Building the unit into a larger project will require some careful planning, as space has to be found for both the circuit board and the loudspeaker. This will normally necessitate using a somewhat larger case than would otherwise be required. The notes on mounting the loudspeaker provided previously also apply here.

It might be possible to add the module into an existing project, but this is dependent on there being sufficient space available in the case. There must also be sufficient front panel space for the loudspeaker. It will be necessary to partially dismantle the project so that the cutout for the loudspeaker and mounting holes for the circuit board can be added without damaging any of the original components.

If there is not enough space available to add the module into an existing project, it might be better to construct it as an external add-on rather than re-housing the project in a larger case. A twin cable plus suitable connectors will then be needed to connect the two units together.

LINKING-UP

In order to use the module as part of a larger project, it is essential to have a certain amount of technical knowledge. It is not possible to provide detailed connection information for a wide range of projects here, and this project is aimed at those who have some experience of electronic design and know what they are doing. In most cases a certain amount of experimentation will be needed in order to get things working well.

If the main project operates from a 5V supply it might be possible to use an output of the project to control the module via pin 23 or pin 24 of IC1. In most cases it will be easier to simply connect it in place of an LED indicator, making sure that it is connected with the correct supply polarity. *The ISD1416P will probably be destroyed if the supply is connected with the wrong polarity.*

The current consumption of the module is likely to be somewhat higher than that of an LED (and can be as high as 30mA). There is no point in trying to use this module with outputs that can only supply a few milliamps.

Sometimes, LEDs are driven from a switching transistor, as shown in Fig.6a. This type of stage can usually provide quite high output currents, and it should control the module without any problems. If necessary the base resistor (R_a) can be reduced in value slightly, but this will not normally be necessary.

Of course, the module should be driven direct from the collector of the transistor, as in Fig.6b, and current limiter resistor R_b should be omitted. If the LED is driven from the output of an integrated circuit via a current limiter resistor (Fig.6c), it will usually be possible to drive the voice module direct from the output of the integrated circuit, as in Fig.6d.

There are likely to be problems if the output is specifically designed to drive an LED, and there is a built-in series resistor or current regulator circuit. Direct control of the voice module is then unlikely to work, and a switching stage will have to be added.

TESTING, TESTING, 1-2-3

Once the project has been completed and all the wiring has been thoroughly checked, it is time to power-up the circuit board and record your message. Pressing switch S1 or wiring pin 27 of IC1 to the 0V supply rail will force the module into a record cycle, and LED D1 should light up to indicate that recording has commenced. If D1 fails to light, disconnect the power at once and recheck the circuit board.

Assuming all is well, speak your message clearly and in a reasonably loud voice. Electret microphone inserts are not usually very sensitive, so you will probably have to be within about 300mm of the microphone in order to obtain good results.

Release S1 or remove the wire link as soon as you have completed your message, which must be no longer than 16 seconds in duration. D1 should then switch off.

In order to play back your recording, either press switch S2 briefly, or remove and reconnect the supply, as appropriate. If the volume is very low and the signal is distorted, it is likely that the microphone insert has the wrong polarity. Reconnecting it with the correct polarity and recording your message again should rectify the problem.

Go to next section