

VP-880 VOICE DEVELOPMENT SYSTEM USER'S MANUAL

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Chapter 1

INTRODUCTIONS

The VP-880 Voice development System is an IBM PC based voice digitization system used to record and organize voice files for EPROM-based QuikVoice modules and ICs. Messages can be either directly recorded from a microphone or downloaded from a cassette/tape player. Here "messages" can be any sounds, including human voice, animal sounds, special sound effects or even music. Messages are actually digitized, not synthesized, so they sound as natural as the original. Once recorded, these messages become "voice files" and are usually stored on a hard disk in the host computer. These voice files can then be downloaded into a standard EPROM programmer and used to program EPROMs.

The VP-880 comes in a "kit" form consisting of a voice digitization board, a speaker, a microphone and a system program. The voice digitization board is an 8-bit add-on card to be installed into any available slot in the host computer. Under proper software control, it basically turns the host computer into a digital recorder. The recording procedure is very similar to the way you use a tape recorder, except that the messages are stored on a hard disk instead of a tape. There are two input jacks on the digitization board: one for the supplied microphone and the other for a line level output device such as a tape player. The supplied speaker connects to the board's output jack for message reviewing purpose.

The system program, with menu-driven user interface, is very easy to use. It provides the following functions:

- Record and play messages
- Test and display current sampling rate
- On-screen message length counter (like a tape counter)
- Save all or part of a message as a separate message
- Combine messages together for multiple-message applications

Voice digitization is a way of turning real-life (analog) signals into computer (digital) format. The process is very similar to filming a movie: you take snap shots of a continuous motion at a fast, constant rate. When the film is played back, the motion is re-constructed and seems to be continuous again. The rate at which this process is done is called the sampling rate. Basically, the sound quality gets better with a higher sampling rate. However, a higher sampling rate generates more data for storage, translating into higher memory cost. Therefore you should select a sampling rate carefully in order to obtain the optimum quality/cost trade-off. The VP-880 allows you to select any sampling rate from 12 Kbps to 64 Kbps by turning a adjustment knob.

There are many different encoding techniques for voice digitization. The VP-880 uses one called CVSD, or Continuously Variable Slope Delta modulation. This technique, encodes the direction of the signal waveform, one bit per sample. Since a bit can only have two possible states, the signal is encoded to be either going up (bit = 1) or down (bit = 0). A period of silence should, theoretically, be encoded as "01010101...". In reality, however, it looks more like "00101110...".

There are both a ADC (Analog to Digital Converter) and a DAC (Digital to Analog Converter) on VP-880. When VP-880 is in the record mode, both ADC and DAC are put to work at the same time. The ADC converts sound input into digital data (to be stored on a disk) while the DAC takes that digital data and converts them back to sound simultaneously. Therefore, when recording, the sound coming out of VP-880's speaker has actually gone thru both ADC and DAC. The sound's quality (at a particular sampling rate) will be very similar to what you will get when it is played back from the actual playback unit.

The VP-880 also incorporates highly sophisticated input and output filters. The input filter limits the input bandwidth (before signals enter the ADC) in order to avoid signal aliasing. The output filter smooths out the rough edges of the the DAC output. There are two sets of filters on VP-880 for two different bandwidths. One set is used for the standard bandwidth (300 to 3.4 KHz) and the other is used for the extended bandwidth (100 to 5.4 KHz). Selection is made via a DIP switch.

Chapter 2

PACKING LIST & SYSTEM REQUIREMENTS

The VP-880 Voice development system comes with the following items (one each):

- (IBM AT compatible) voice processing board
- system program diskette
- microphone
- speaker
- line input adaptor cord
- User's Manual

It requires a host computer (not included) with the following minimum specifications:

- IBM 386 compatible personal computer, 25 MHz or faster
- minimum 512K system memory (EMS memory highly recommended)
- a 3.5" or 5.25" floppy disk drive
- a hard disk is highly recommended but not required
- all video display types are supported
- PC-DOS or MS-DOS, version 3.0 or higher

Chapter 3

INSTALLATION AND TESTING

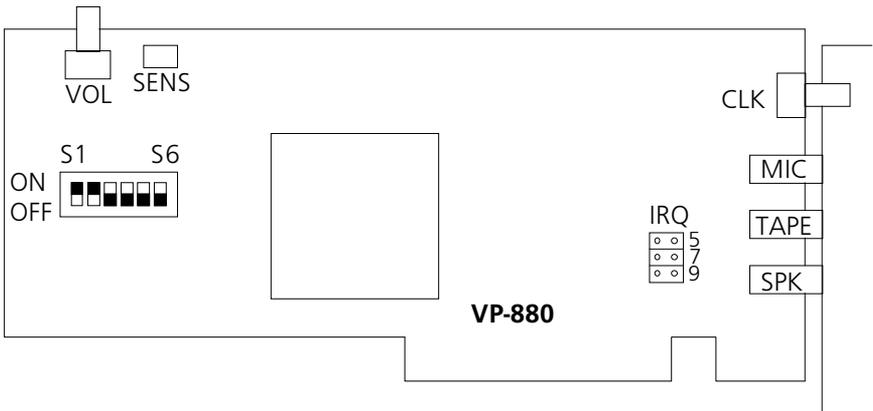
The VP-880 digitization board can be installed into any available slot in the host computer by following these easy steps:

INSTALLATION STEP 1:

VP-880 can use interrupt IRQ2 (default) 5 or 7, but it always uses I/O address 238H to 23FH (eight bytes). Check all add-on boards in your computer and make sure there are no conflicts. Pay special attention to devices such as bus mouse, network interface card and VGA card, they may use IRQ2 (same as IRQ9). If there is an IRQ conflict, change the VP-880's IRQ setting by moving the IRQ jumper to another location. Please refer to the circuit board diagram below.

INSTALLATION STEP 2:

Select the proper filter bandwidth on the VP-880 board by setting the 6-position DIP switch as following:



DIP Switch Setting	Filter Bandwidth	Suitable Sampling Rates
S1=S2=ON, others=OFF	300 to 3.4 KHz	12 to 64 Kbps
S1=S2=OFF, others= ON	100 to 5.4 KHz	64 to 128 Kbps

This selection should be made based on your sampling rate and the nature of your sound. If you select the extended bandwidth (100 to 5.4 KHz), you WILL have more background noise in your message, due to the wider bandwidth. If you are not sure, you can always change it later.

INSTALLATION STEP 3:

Turn off the host computer. Install VP-880 board into any available slot firmly but leave the cover open for now. You may need to make some manual adjustment later.

INSTALLATION STEP 4:

Connect the microphone and the speaker to their respective jacks on the VP-880 board. Refer to the following diagram for jack locations.

After the board is installed, turn on the host computer and install the program with these steps:

INSTALLATION STEP 5:

Make a backup copy of the supplied diskette and put it in a safe place. The diskette is not copy protected. Then use the DOS "COPY" command to copy all files from the diskette to your hard disk. We recommend that you create a subdirectory for storing VP-880 files only. For example:

```
MD VP880                (create a subdirectory called VP880)
XCOPY A:*.* VP880 /S   (copy all files from drive A to VP880)
```

The installation is now completed. The following steps are used to test the board to see if the installation is OK.

TESTING STEP 1:

From the directory where the program is stored, enter "VP880 /IRQ=?" where "?" is the IRQ setting (2, 5 or 7). If you followed the example in step E above, you will have to change directory first:

CD VP880 (change to the VP880 subdirectory)

A menu should come up shortly. Look for a blinking sampling rate display on the lower left part of the screen. If it shows 0.0 Kbps, then the board is either defective or not installed properly (possibly IRQ or I/O conflict). Otherwise, carefully turn the sampling rate knob (VR3) and change the sampling rate to about 32 Kbps. It may be hard to reach exactly 32.0 Kbps, but anything within the +- 1 Kbps range is acceptable. Human ears can hardly tell the (pitch) difference due to such a small deviation.

TESTING STEP 2:

Select the "Load" function from the on-screen menu. Follow the instructions and load the demo message file "880DEMO.VOC".

TESTING STEP 3:

Select the "Play" function. Follow the instructions to play the demo message. If you can't hear the message, try adjusting the speaker volume by tweaking the volume control knob labeled "VOL". It will also help if you double-check the speaker cable and make sure it's firmly connected to the board. If the message sounds OK, proceed to the next step.

TESTING STEP 4:

Select the "Record" function. Turn on the microphone (remember to put in the battery first) and start speaking. You should hear voice of yourself coming from the speaker clearly. If not, try adjusting the microphone sensitivity by turning the sensitivity pot labeled VR1. If you still can't hear anything, test the microphone elsewhere and make sure it's good. If you can hear yourself but the sound quality is poor, it may be due to

- (1) the microphone sensitivity needs adjustment, or
- (2) the microphone needs a new battery, or
- (3) the board is defective.

You may adjust the sampling rate while you are recording to hear the sound quality at different sampling rates. If 32 Kbps does not sound good enough you may increase the sampling rate. Note that the sampling rate display will not be updated until you do another "Sampling Rate Test".

If the board passed all the tests described above, the system is properly installed and ready to use.

Chapter 4

RUNNING THE SYSTEM PROGRAM

The system program is menu-driven and very user-friendly (ha ha ha). To start the program, first change to the subdirectory where the program is stored, then enter "VP880 /IRQ=?" or "880EMS", where "?" is the IRQ setting. Note that "880EMS" supports IRQ=2 only. These two programs are basically the same except that "VP880" uses convention DOS memory while "880EMS" uses EMS memory. If you are going to record messages longer than 500K bytes (single message length, not combined) you must have EMS memory in your computer and use "880EMS". Otherwise, "VP880" is adequate and more efficient than "880EMS". If you need help on EMS memory, see Appendix A.

Once the proper program name is entered, the screen displays a menu similar to the following:

```
----- Eletech Electronics VP-880 Speech Development Program Version 1.2 -----
+-----+-----+-----+
|          MAIN MENU          |          RECORD / PLAY          |          ROM DATA MANAGEMENT    |
+-----+-----+-----+
| [A] Sample rate Test        | Memory Used :00000000 | Segment ( 1 ) : 0000000 |
| [B] Speaker ON/OFF         | Current Byte : 0000000 | Segment ( 2 ) : 0000000 |
| [C] Record                  | Total Memory :00510352 | Segment ( 3 ) : 0000000 |
| [D] Play                    | +-----+          | Segment ( 4 ) : 0000000 |
| [E] Load                    | +-----+          | Segment ( 5 ) : 0000000 |
| [F] Save                    |          SPEECH EDITING    | Segment ( 6 ) : 0000000 |
| [G] Speech Editing          | +-----+          | Segment ( 7 ) : 0000000 |
| [H] ROM Data Management     | Start Address: 0000000 | Segment ( 8 ) : 0000000 |
| [ESC] Quit                  | Stop Address : 0000000 | Segment ( 9 ) : 0000000 |
|                               | Step Size   : 0000512 | Segment (10) : 0000000 |
|                               | Window Size : 0000000 | Segment (11) : 0000000 |
| (c) Eletech Electronics     |                               | Segment (12) : 0000000 |
| 1990 All Rights Reserved    |                               | Segment (13) : 0000000 |
+-----+-----+-----+
|                               | [ ] Start Address      | Segment (14) : 0000000 |
|                               | [ ] Stop Address       | Segment (15) : 0000000 |
| SAMPLE RATE : 32.0 Kbit/s   | [A] Set Start Address  | Segment (16) : 0000000 |
|                               | [B] Set Stop Address   | Segment (16) : 0000000 |
| SPEAKER STATUS : ON         | [C] Set Step Size     | Total Length : 0000000 |
|                               | [D] Save Data To Disk | ROM Size      : 0000000 |
+-----+-----+-----+
```

When the program is first activated, it checks the VP880 board and flashes the current sampling rate on screen. If the sampling rate shows a constant "0.0 Kbps", the board is not working properly. Refer to Chapter 5: Solutions to Common Problems for solutions. Otherwise, you may start adjusting the sampling rate (see A. Sample Rate Test for details) or press any key to stop the flashing and begin normal operation.

The screen display is partitioned into the following windows:

Main Menu

Contains a list of available functions

Record/Play Menu

Contains system memory information

Speech Editing Menu

Contains a list of available commands for speech editing

Rom Data Management Menu

Contains segment information for multiple message

Sample Rate & Speaker Status Display

Shows the sampling rate (Kbps) and speaker status (ON or OFF)

Main Menu functions are described below. Other menus and displays are also described below at appropriate points.

A. Sample Rate Test:

The purpose of the Sampling Rate Test is to provide a continuously updated sampling rate display on the screen so that you know where you are when making sampling rate change. You may turn the sampling rate knob at any time to change the sampling rate, but only in the Sampling Rate Test will you get an instant feedback of the current sampling rate.

To adjust the sampling rate, choose the Sample Rate Test from the main menu by pressing <A>. The on-screen sampling rate display should become flashing. Gently turn the sampling rate knob (labeled "CLK", or the only knob on the metal bracket) clockwise to increase the sampling rate, counter-clockwise to decrease the sampling rate. The sampling rate

display will change instantly as you turn the knob. Press any key when you are done.

Note that the sampling rate knob is very sensitive and it could be hard to adjust the sampling rate to exactly what you want. However, a tolerance of ± 2 Kbps is usually acceptable since it has very little effect on the sound quality. Sometimes the sampling rate will show 0.0 Kbps momentarily, especially when you are adjusting it. This is OK.

If you have turned the sampling rate knob since the last time you ran the Sample Rate Test, VP-880 is probably running at a different sampling rate than the one shown on screen. Although a wrong sampling rate display does not affect the normal operation of VP880, it is misleading and should be avoided.

B. Speaker ON/OFF:

This function toggles the speaker on and off. The speaker status is always displayed on the screen in the "SPEAKER STATUS" field. Most of the time you want to leave the speaker on so you can hear the sounds. But when you use a microphone to record messages, you may want to turn off the speaker in order to avoid the (possible) hawling effect (speaker producing a loud, high pitch noise due to positive feedback). If you are transferring messages from a tape player into the system via the LINE input jack, you may safely turn on the speaker without causing the hawling effect.

To turn the speaker on or off, simply press from the main menu. The "SPEAKER STATUS" display will toggle between ON and OFF to indicate the current speaker status.

C. Record:

Before you do any recording, make sure you have adjusted the sampling rate to the desired setting and selected the proper filter bandwidth. Refer to Chapter 3: Installation and Testing for details on these adjustments.

Press <C> from the main menu to choose this function. The system will enter the record-standby mode. In the record-standby mode, the sound source (microphone or tape) is looped back to the speaker in real time without doing any actual recording. If the speaker is on, you can hear the sound source which has actually gone through the entire digitization process (encoding and decoding). Therefore the sound quality will be very similar to what you will get in the actual application.

Once in the record-standby mode, the following options are available:

- | | |
|------------------------------|---|
| <Esc> Key | Quit this function |
| <A> Key | Append to a previous recording |
| <Any Other Key> | Start a new recording and overwrite system memory |

When recording, the message will be digitized and stored temporarily in the system memory. Thus the available system memory amount limits the maximum length of a single message. If you have EMS memory and run "880EMS.EXE", the entire EMS memory can be used to store the recorded data. This usually means that you can record 2 or 3M bytes of data as a single message. If you don't have EMS memory, you will have to run "VP880.EXE" and be limited to about 500K bytes maximum length for any single message.

In the Record/Play Menu, the Current Byte field counts up as recording goes on, showing you how many bytes have been recorded. If you start a new recording, it counts up from zero. If you append to a previous recording, it counts up from the end of the previous recording. The Memory Used field shows the amount of memory used so far. The Total Memory field shows how much memory (in number of bytes) is available.

To stop recording, press any key. You will be returned to the Main Menu. You may then:

- Play the message to see if it's OK (use the "Play" function), or
- Save the message to your hard disk (use the "Save" function), or
- Edit the message (use the "Speech Editing" function), or

- Append to the recording (use the "Record" function), or
- Discard the recording and start a new one (use the "Record" function)

If you want to append to a message file already stored on your hard disk, you may do so by first loading it into the memory (use the "Load" function) and then append to it (use the "Record" function).

To obtain the best sound quality for a certain sampling rate, first have the message recorded by a voice talent in a recording studio with professional equipment onto a cassette tape. Then download the message into the system via the TAPE input jack. No recording level (sensitivity) adjustment is necessary in this case.

If you are recording directly from a microphone, you will definitely get more background noise. You should adjust the microphone sensitivity pot on the VP-880 board to reach a proper recording level in order to avoid clipping distortion. There is no VU meter available for this purpose, so you have to do some "try and error" experiment until you get it right. The rule of thumb is that your recording should sound about as loud as our demo message ("880DEMO.VOC").

If you want to record several messages in one shot, you may do so as long as you have enough memory to buffer them. For example, if you have five separate messages in a roll on a cassette, you may record them as a single message and then separate them by using the "Speech Editing" function. Once separated, you may save them to the hard disk as separate files.

The "Speech Editing" function also allows you to trim off the dead space in the recording (undesired silence before and/or after the actual message). See its description in this chapter for details.

Finally, always remember to save the recording after you have done the necessary editing and/or separation. If you don't save it, it will not be stored on the hard disk for later use.

D. Play

This function is used to play the message stored in the system memory. If you want to play a file stored on your hard or floppy disk, you must load the file into the system memory first by using the "Load" function. If you have just made a new recording, it is already in the system memory and you can play it without doing any loading.

Press <D> and follow the instructions on screen to play. As the playback goes on, the "Current Byte" field in the Record/Play Menu will count up, showing you the number of bytes currently being played. If you don't stop the playback with any key press, it will play the full length specified in the "Memory Used" field and stop by itself.

E. Load

Press <E> to load a sound file from a hard or floppy disk into the system memory, for playback or editing purpose. You may specify a full path name such as "A:TEST.VOC" or "\\VOICE\TEST.VOC". Note that once the file is loaded into the system memory, previous memory contents are totally lost. So remember to save your recordings with the "Save" function before loading a new file.

After the file is successfully loaded, the "Memory Used" field in the Record/Play Menu will show the file length in number of bytes.

F. Save

Press <F> from the main menu to save your recordings to disk. The amount of memory saved is always the amount shown in the "Memory Used" field in the Record/Play Menu. The file format is "Binary", which basically means that the file contains only data bytes with neither header information nor checksum bytes. You must use this function to save your recordings or they will be lost when you exit the program. However, this function is not the only function that saves files. There is also a "save"

command in the "Speech Editing" function, which allows you to save a specific portion of the memory.

If you don't want to save the file in the current directory, you may specify a full path name such as "A:\BACKUP\VOICE\TEST.VOC" or "FILES\TEST.VOC". You may save files to a hard disk or a floppy disk. If the file already exists, you will be asked to either overwrite it or select another filename.

After saving, the system memory is not cleared and still contains the same data as before. The old data will be lost only when you load a new file, make a new recording, or exit the program.

G. Speech Editing

Pressing the <G> key from the Main Menu puts the system in the Speech Editing mode. In this mode you can select and save a portion of the system memory. If you want to edit a message file stored on a hard or floppy disk, you have to use the "Load" function to load it into the system memory first.

In the Speech Editing mode, the following commands are available:

<Up Arrow>, <Down Arrow> Moves the Start Address by a Step Size

<Left Arrow>, <Right Arrow> Moves the Stop Address by a Step Size

<A> Type in the Start Address directly

**** Type in the Stop Address directly

<C> Change the Step Size

<D> Save data from Start Address to Stop Address to disk

<Space Bar> Play data from Start Address to Stop Address

Using the above commands you will be able to "cut" a section of the recording and save it as a separate file. If you want to "paste" sections of a message together, you may use the DOS "COPY" command as following:

COPY /B file1+file2+... file_all

This command copies file1, file2... together into one file "file_all". However, this operation is not the same as the file assembling operation provided in the ROM Data Management function. The DOS "COPY" command simply copy all files together. It does not check for and eliminate EOM flags and it does not patch up the destination file with "silence" to match the EPROM size neither. Therefore you should use the "COPY" command to combines sections together into a single message, but not to combine messages together for multi-message applications.

The Speech Editing function makes it possible for you to record a number of messages at once and separate them later. For example, if you record three messages sequentially into the system memory, you may separate them by taking the following steps:

- a. Select the "Save" function and save the recording as a backup copy.
- b. Select this function and play the recording. Watch the message counter counts up while the recording is being played. Write down the counter value when the first message starts to play, and also when it finishes. This is the start and stop address of the first message. Set the start and stop address of the buffer window accordingly and save it.
- c. Repeat step B for the other two messages.

If these three messages are to be used for a multiple-message voice module or IC, you have to use the "ROM Data Management" function to combine them after they are separated into individual message files.

If these messages are to be used for single-message voice modules, then no other operation is required. Just download the message as a binary file to an EPROM programmer and program EPROMs.

H. ROM Data Management

This function is used to combine several message files into a multiple-message format. It adds the necessary "End Of Message" (EOM) code between messages so that multiple-message voice processors (VP1410 and VP1610) knows how to locate the target message.

Press <H> to enter the ROM Data Management Menu. The operation is pretty straight forward. First you tell the system what size of EPROM you are using. Available selections are 256K, 512K, 1M, 2M, 4M and 8M. Then, when the Filename Menu opens in the center of the screen, you enter the filenames of the messages to be combined, in the order you want them to appear in the EPROM. Length of each file (in bytes), along with the combined length and number of EPROM needed, is displayed for your reference.

You may enter filenames with search path. If you enter a filename which can not be found, the computer will beep and refuse to go on. You have to enter a correct filename to continue or press <Esc> to quit.

The following commands are available to help you enter the correct filenames:

<Up Arrow>, <Down Arrow> Moves the cursor up and down.

<F1> Search and select files.

The command allows you to view the contents of a directory (similar to a DOS "DIR" command) and select files without typing in filenames. After you press the <F1> key, a small window opens up on the left side of the screen, which will be used to display directory contents, and the "File Name" window in the center is the one that you use to enter filenames. On the bottom of screen it says:

Please Input Pathname : *.*

At this point you have to either press the <Enter> key to accept the default path (*. * = all files in the current directory) or enter a path name such as "C:\VOICE*.VOC". The "File Name" window on the left will then be filled with filenames found in the path. You have the following options:

<Up Arrow>, <Down Arrow> Move the cursor up and down.

<Ins> Mark the file.

**** Un-mark the file.

<Space Bar> Play the file without marking it.

<Enter> Quit and select all marked files. This command brings all marked filenames into the "File Name" window.

<Esc> Quit without selecting any files.

<F2> Assemble files.

After entering all necessary filenames, you are ready to press <F2> to assemble them together. The assembling process does the following operations:

1. Scan through every source file and eliminate all occurrence of the EOM flag within each file. The EOM flag consists of six consecutive bytes of AA ("10101010") and could appear occasionally in an un-assembled file. When an undesired EOM flag is found in the middle of a file, one of the bits will be changed so that it's no longer a EOM flag.

2. Add a EOM flag to the end of every source file and combine them together into a single, temporary file. The EOM flag is used by multi-message voice processors such as VP1410, VP1606 and VP1608 to locate the proper message in an EPROM.

At this point you will be prompted (at the bottom of screen) to either play the temporary file (press <1>) or "Page & Save" it (press <2>). The "Page & Save" command breaks the single, temporary file into several smaller, target-EPROM sized files. If the temporary file is smaller than the target EPROM in the first place, it is simply patched with enough dead space to increase its size to match the target EPROM's.

Before storing assembled files to disk, VP-880 asks you to enter a filename with a search path (if any) but no extension. It automatically adds a number as the filename extension to the files. For example, if three files are created as a result of assembling eight individual message files, they will be named "FILENAME.1", "FILENAME.2", "FILENAME.3". These binary-format files are of the same size as the target EPROM and can be downloaded to a standard EPROM programmer to program EPROM chips. Remember to use the "Binary" format when downloading the files to the EPROM programmer. Refer to your EPROM programmer's manual for downloading procedures.

Chapter 5

SOLUTIONS TO COMMON PROBLEMS

All of the following problems could be due to a defective VP-880 board. If, after trying all suggested solutions below, you still can not make it work, then contact your dealer for further assistance.

PROBLEM #1: The sampling rate shows 0.0 Kbps.

It's OK if the sampling rate shows 0.0 Kbps momentarily when you are adjusting it. But if it shows 0.0 Kbps constantly, the board may not be working properly. There is probably an IRQ (more likely) or I/O address (less likely) conflict between the VP-880 and another add-on board in your system. If you can not determine which one is causing the conflict, try removing all add-on boards except the video card and the floppy/hard disk controller card. If doing so solves the problem, add those add-on boards back one by one and see which is causing the problem.

Once you find out which add-on board is conflicting with the VP-880, you must re-configure its IRQ or I/O address (whichever is causing the conflict). Note that VP-880 must use IRQ 2 and I/O address 238H.

Another possibility is that the bus speed of your computer is too high. If your computer has a bus speed higher than 8 MHz, try lowering it to 8 MHz or slower. This can usually be done by changing the "advanced CMOS setup" in your computer. Do not confuse the bus speed with the CPU speed. The VP-880 can work with any CPU speed and, actually, the higher the better.

If the VP-880 still shows 0.0 Kbps sampling rate, the board may be defective. Contact your dealer for repair or replacement.

PROBLEM #2: Nothing comes out of the speaker.

1. If the sampling rate reads 0.0 Kbps, see PROBLEM #1 for solutions.
2. Check the speaker connection. Note that there are three identical jacks on the VP-880 board, and the speaker jack is the bottom one.
3. Make sure the speaker is turned ON in the VP-880 program.
4. Adjust the speaker volume. The volume knob is mounted on the VP-880 board and not accessible from outside the computer, you have to open the computer to make the adjustment. See the "VP-880 Board Diagram" in Chapter 3 for the location of the volume knob. Turning the knob clockwise increases the volume, counter-clockwise decreases it.
5. If you are using a microphone, make sure it's switched on. Check the battery inside the microphone if there is one. Test the microphone elsewhere and make sure it's good.
6. Test the speaker elsewhere and make sure it's good.

PROBLEM #3: How do I install EMS memory in my computer?

First, be aware that you don't need any EMS memory if you are not going to run "880EMS.EXE" of which the only benefit is allowing you to record a message longer than "VP880.EXE" can handle (usually larger than 4M bits).

Second, having EMS memory in your system will add additional burden to your computer. This means that if your computer is not that powerful in the first place, EMS memory will make it worse.

If you have a 8088- or 80286-based PC, you should replace it with a 80386 or better computer. Equip the computer with at least 4MB of memory.

Upgrade the DOS to version 5.0 or higher (if applicable). Add the following two lines to your CONFIG.SYS file (if they are there already, then you already have EMS memory):

```
DEVICE=C:\DOS\HIMEM.SYS  
DEVICE=C:\DOS\EMM386.EXE ????
```

Here "?????" specifies how much memory (in kilo-bytes) should be turned into EMS memory. If you have 4MB of memory and no other driver is using it, you may try "2800". If everything goes well you should see a message on the screen (when the computer is booting up) telling you how much EMS memory has been installed. Otherwise an error message will be displayed. If you encounter any difficulties when installing the EMS memory, refer to your DOS manual or consult with a computer expert.

PROBLEM #4: All my recordings are silent or very weak, but the demo message played fine.

If you use a microphone to record messages:

1. Make sure the microphone is switched on and the battery inside (if any) is good.
2. Adjust the microphone sensitivity pot and try again.
3. Check the cable connection. The microphone jack is the top one.

If you use a tape player to transfer messages:

1. Check the cable connection. The tape jack is the middle one.

PROBLEM #5: Sound quality is bad.

It is hard to define "bad" quality. A little bit of background noise is normal with digitized voice. If there is excessive background noise then (1) the recording level may be too low, (2) the host computer has a noisy power supply, or (3) the board might be defective. If there is a popping noise from time to time, especially when the disk is being accessed, it's most likely due to the disk drive being too slow.

If the sound pitch is apparently lower than normal, the host computer is probably too slow. If the sound is distorted, then the recording input level may be too high (signal waveform is clipped on the top). One last question, is the original sound source clear in the first place?

PROBLEM 6: How can I improve the sound quality in my actual playback device?

Besides increasing the sampling rate, try the following:

1. Use a graphical equalizer to pre-condition your source sound. For example, if you need more bass, simply boost the low frequencies with the equalizer. This method proves to be very effective.
2. Select the proper filter bandwidth on the VP-880 board by setting the 6-position DIP switch as following:

DIP Switch Setting	Filter Bandwidth	Suitable Sampling Rates
S1=S2=ON, others=OFF	300 to 3.4 KHz	12 to 64 Kbps
S1=S2=OFF, others= ON	100 to 5.4 KHz	64 to 128 Kbps

This selection should be made based on your sampling rate and the nature of your sound. If you select the extended bandwidth (100 to 5.4 KHz), you WILL have more background noise in your message, due to the wider bandwidth. If you are not sure, digitize your message with both bandwidths and see (or hear) which one you like better.

3. Use a proper speaker in your application. A good speaker will certainly make a lot of difference in the sound quality. Also, a speaker in an enclosure will usually sound better than a speaker without an enclosure. If you don't get enough volume output from your speaker, try using one with a bigger physical size. Sometimes sound quality is degraded when an under-powered speaker is used.

PROBLEM 7: Message files are not assembled correctly.

Your hard disk or floppy disk is probably almost full. In order to assemble message files, VP-880 needs a working disk space of roughly the same size as the files combined. If there is not enough available disk space, VP-880 will produce files of erroneous size (without a warning!)

Appendix A:

HARDWARE SPECIFICATIONS

Voice Input:

JR1 - microphone input (3.5mm (1/8") phone jack)

JR2 - line level input (3.5mm (1/8") phone jack, mono or stereo)

(Stereo signals will be mixed into a mono sound for digitization)

Input Filter Bandwidth: 300 Hz to 3.4 KHz (standard) or

100 Hz to 5.4 KHz (extended)

Digitization Technique: CVSD with variable sampling rate

Voice Output:

JR3 - speaker output (3.5mm (1/8") phone jack)

Manual Adjustments:

VR1 - microphone sensitivity

VR2 - speaker volume

VR3 - sampling rate (12 to 64 Kbps)

Host Computer Interface:

Bus Interface: IBM PC ISA 8-bit

I/O Address: 238H to 23FH

Interrupt Request: IRQ 2

DMA Request: none

Voice Compression: none

Most voice compression schemes compress voice data at the cost of sound quality. For example, the ADPCM technique usually boasts a compression ratio of 2:1 or higher. In fact, a so-called "4-bit ADPCM" usually samples in 8-bit (resolution = 8 bits), but stores only 4 bits of data which encode the change of signal value from the previous sample. If the

change is bigger than 4 bits can encode, distortion occurs. The CVSD technique that we use is sort of a "1-bit ADPCM". Each sample value change is encoded in just one bit.

In order to compensate for the smaller number of encoding bits, CVSD is usually sampling, in terms of KHz, at a much higher rate than an ADPCM. This means that CVSD can usually achieve the same compression result as the so called voice compression techniques by just reducing its sampling rate. Because both ways you get less amount of data with a lower sound quality. We dare to compare the sound quality of our modules and ICs with any other brands at the same data rate, compressed or not.