ICs for Speech Synthesis

OM8210 User's Manual (IBM PC - XT - AT)

Laboratory Report

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Abstract: The Speech Adapter Box (SAB) together with the OM8210 speech processing software and an IBM-PC/XT/AT form a complete stand-alone speech coding system for the Philips PCF8200 speech synthesizer.

Human speech can be processed from audio tape via sampling, analysis, parameter editing, and coding to writing EPROM's that fit your specific speech application hardware.

OM8210 User's Manual (IBM PC - XT - AT)

Authors: H. Zelle/M. Roberts

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Central application laboratory CAB Eindhoven – The Netherlands

SUMMARY

The Speech Adapter Box (SAB) together with the OM8210 speech processing software and an IBM-PC/XT/AT (or compatible) with a minimum memory size of 400 kbyte and extra cards for IEEE488 interface and graphics make up a complete stand-alone speech encoding system for the Philips PCF8200 speech synthesizer.

Human speech can be processed from audio tape via analysis and editing to EPROM's for your specific speech application hardware.

Many commands are available creating a flexible speech editing system. The interactive response of the system, the "confirm" protection of destructive commands and the always available explanation of the command characters make it easy to use.

The reader of this manual is assumed to be familiar with the fundamentals of human speech and the formant synthesis model used in the PCF8200 (see: Elcoma Technical Publication 217).

CONTENTS

1. INTRODUCTION

The Speech Adapter Box (SAB) together with the OM8210 speech processing software and an IBM-PC/XT/AT (or compatible) with a minimum memory size of 400 kbyte and extra cards for IEEE488 interface and graphics make up a complete stand-alone speech encoding system for the Philips PCF8200 speech synthesizer.

Human speech can be processed from audio tape via analysis and editing to EPROM's for your specific speech application hardware.

Many commands are available creating a flexible speech editing system. The interactive response of the system, the "Y/N" protection of destructive commands and the always available explanation of the command characters make it easy to use.

In this manual the following syntax is used to describe user input:

```
[...]          = key to be pressed
[...]+[...]    = keys to be pressed simultaneously
[....]n        = key followed by number input
"....."        = text to be entered litterally
(...)          = parameter to be specified by user
{......}       = optional parameter.
```

Number input is terminated with [Enter] or [space].
In the description of user input it is assumed that "Capslock" mode has been activated.

The reader of this manual is assumed to be familiar with the fundamentals of human speech and the formant synthesis model used in the PCF8200 (see: Elcoma Technical Publication 217).

2. CONFIGURATION OF THE COMPUTER

The OM8210 speech software for the IBM-PC/XT/AT has been developed under DOS 3.30 for the following hardware configuration:

- IBM-PC/XT/AT with monochrome display and at least one floppy disc drive suited for reading 5¼ inch diskettes of 360 kbytes
- At least 400 kbyte RAM memory for 4 seconds of speech (each extra 24 kbyte of memory gives 1 additional second of speech processing capacity for the PCF8200)
- Tecmar IEEE488 board (rev. D) or the hardware compatible Scientific Solutions IEEE488 (rev. A)
- Hercules monochrome graphics card (720x348 dots; rev. C)

The software also runs on a Philips P3100 personal computer (see Chapter 17: Option).
For other configurations or compatible machines the software has not been tested and may not run properly.

3. INSTALLATION

First you should unpack and install your IBM or compatible machine according to its installation manual. The video display unit, however, should not be connected to the normal socket but to the Hercules graphics card.

The switch settings of the IEEE488 board can be found on page 17 of the IEEE488 installation manual (both for Tecmar and Scientific Solutions). Figure 2 on that page gives the factory setting of the switches to make the board an IEEE488 system controller with I/O address starting at 0310 Hex. (SW1 = 11111111; SW2 = 11111111; SW3 = 01110011; SW4 = 11; SW5 = 11110011; JMPRs 1,2,3,4 not connected).

The IEEE-to-IEC bus adapter (PM 9483/50 or PM 2296/50) should be mounted on the IEEE488 connector on the back of the computer. This may present some mechanical difficulties on non-IBM machines. Use the supplied flat cable to connect this adapter to the SAB IEC625 connector.
Upon delivery, the SAB device address is set to 19. Make sure that no other device with that address is connected to the IEEE bus. If this cannot be avoided, change the SAB hardware and software device address (see Chapter 17: Option)

The SAB power supply (PE 1140/50) is factory set for 220 VAC. It can be changed to 110 VAC (see Appendix D).

4. SAB SOFTWARE AND MEDIA

The OM8210 software (version 1.1) is supplied on a dual-sided 5¼ inch diskette and includes:

- IBMBIO.COM and IBMDOS.COM hidden files
- COMMAND.COM DOS command interpreter
- AUTOEXEC.BAT startup file
- CONFIG.SYS to prepare the I/O system
- ANSI.SYS ASCII keyboard driver
- MODE.COM to set the serial port modes
- DISKCOPY.COM to make backup copies of floppy discs
- FORMAT.COM to prepare new disks
- HGC.COM to initialize the Hercules graphics card
- DOSTIME.EXE to copy time and date from the battery clock (Tecmar Captain Card)
- SETTIME.EXE to set the battery date and time (Tecmar Captain Card)
- SC.EXE to switch from "graphics" screen to "text" screen
- SP2.EXE the speech program
- CONFIG2.PCF the speech program configuration file

Note: Do NOT write-protect the OM8210 diskette! This will cause a runtime DOS error when booting because the configuration file CONFIG2.PCF cannot be rewritten.
Remedy: remove write-protect sticker and press [R] to retry or press [Ctrl]+[Alt]+[Del] for a system reset.

4.1 Data storage

Your IBM PC is equipped with one (or two) 5¼-inch floppy disc drives called "A": (and "B:") and possibly a harddisc unit labeled drive "C:".

At startup the OM8210 diskette must be loaded in drive A. In this way the DOS environment is loaded from the OM8210 disc together with the speech program.
Ater program initialization drive A may be used for speech file storage. Before accessing the configuration file CONFIG2.PCF in the Option mode - using the G(et_conf) or K(eep_conf) command - the OM8210 system disk should be loaded first.

Speech data files have two different formats:

- parameter files, containing an ASCII representation of the frames in the parameter "edit" memory
- code files, containing an ASCII representation of the contents of the "code" memory in INTEL HEX format.

Filenames are composed of the filetype ('P' or 'C') followed by a 3-digit number: e.g. P003 or C192.

A speech file directory (called FDIR) is used to access the speech data files. It contains the date of creation, the length, and a description of each file.
Before a diskette can be used to store speech data files such a speech file directory must be created in the File mode using the M(ake) command (see Chapter 16).

4.2 Disk initialization

All new diskettes must be initialized before they can be used for storage of speech data. This is done by means of the DOS "FORMAT" command, as follows:

"FORMAT (target_drive):(/V)(/1)"[Enter]

The optional parameter "/V" permits the user to enter a volume label to uniquely identify a diskette.
The optional parameter "/1" makes the target diskette single-sided.

The (main) root directory can only contain 64 or 112 file entries (single or dual-sided). This severely limits the possibilities for speech file storage. Therefore a subdirectory must be created which has no limit on the number of files except the disc size. This is done with the DOS "MKDIR" (or "MD") command, as follows:

"MKDIR (subdir_name)"[Enter]

Finally the speech-file directory FDIR must be created to permit speech file access from the OM8210 software. This is done in the File mode of the speech program: first use the P(ath) command to set the pathname to the subdirectory in question, then use the M(ake) command to create a speech-file directory of the required size. See Chapter 16 for more details.

4.3 Making disc backups

Backups of floppy discs can be made using the DOS "DISKCOPY" command, which includes formatting (if necessary) of the target diskette.

"DISKCOPY (source_drive): (target_drive): "[Enter]

In this way the entire diskette is copied track by track. This method is advised for OM8210 system disc backups and backup copies of entire speech data discs.

When the contents of a speech data diskette have become fragmented due to repeated deleting and creating of files, a "clean" backup copy can be made using the DOS "COPY" command and a newly formatted target diskette.

"COPY (source_drive):*.*(target_drive):"[Enter]

All files are now copied sequentially on the target diskette.

5. SAB HARDWARE

Figure 1 shows a block diagram of the speech coding system. All SAB functions are controlled via the IEC/IEEE bus:
input level, output volume, analog multiplexer select, sampling frequency, EPROM type select, etc.

There are five types of data streams to and from the SAB:

1. Hardware control.

2. Audio input from tape via level control, anti-aliasing filter and the A/D-converter.

3. Audio output via the DA-converter, analog multiplexer, LP-filter, volume control and power amplifier.

4. Synthesizer output of parameters/codes via analog multi- plexer, LP-filter, volume control and power amplifier.

5. Input/output of EPROM bytes.

When neither the synthesizer nor the D/A-converter is used, the tape input signal is routed directly to the power amplifier to enable tape input monitoring.
Data streams 2 and 3 are used in the Sample mode.
Data streams 3 and 4 provide direct audio feedback in the Parameter mode and the File mode.

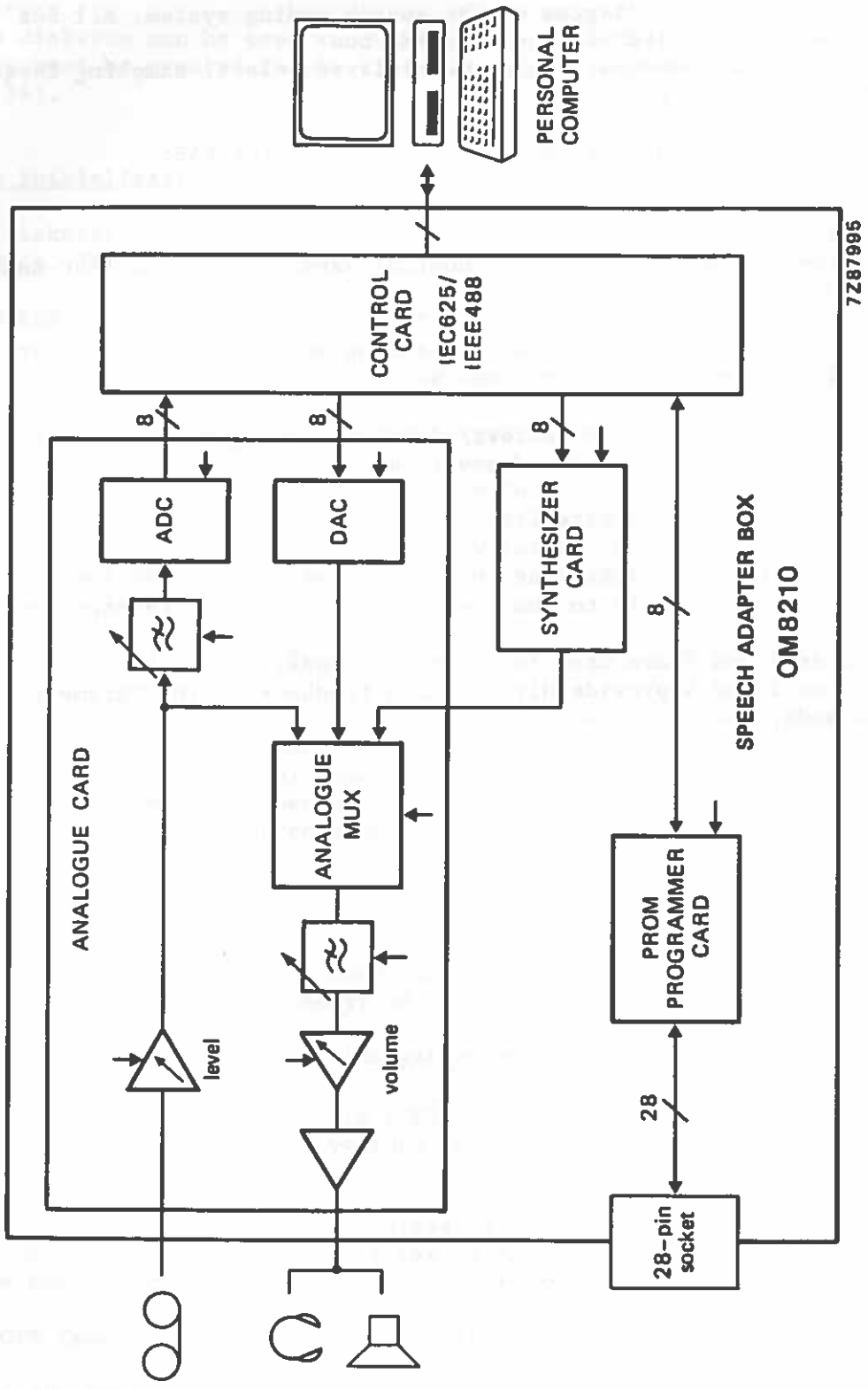

Figure 1. Speech coding system.

6. RECORDING SPEECH MESSAGES

As with the reproduction of all audio signals, the quality of synthesized speech will depend on the quality of the weakest link in the chain from source to listener.
In resynthesizing human speech with a limited model of the vocal tract this is particulary relevant: The analysis software will adjust the filter model to optimally represent the original signal. When the original also includes sounds not originating from a single human voice, the analysis will also try to model these disturbances. It is obvious that the results will then be less than optimal.

Ten Commandments for speech recording

I Use a hifi cassette deck with dolby and high-quality tape (e.g. ferro chromium or metal). S/N ratio must be at least 60 dB.
Alternative: open reel deck at 19 cm/s (7.5 ips).

II Recordings should be made in a quiet non-reverberant room using a hifi unidirectional microphone at a distance of approximately 10 cm from the speaker's mouth. To eliminate the low frequency boost caused by "close-miking" it is advised to use a first-order high-pass filter (f_{co}=200 Hz) or a preamplifier with its bass control set to minimum.

III Speak fairly loudly, as if addressing a small group of people. Adjust the recording level so that the level indicator just exceeds 0 dB VU during the loudest passages.

-------MOST IMPORTANT-------

IV Try to speak all syllables at the same level. Check that the weakest ones reach at least -10 dB on the recording level indicator. Note: this will require some training.

V Use normal rate of speaking. Do not draw out or shorten syllables. Do not emphasize syllables unless required specifically.

VI Record all speech at least twice. Keep checking the level indicator. When in doubt record again: tape is not expensive, delay is.

VII Record isolated words and lists of words with a flat intonation. Do not change intonation within a list of words. Adding an extra word at the end of the list helps to keep the intonation constant.

VIII Use several speakers to record the speech material. Successful coding depends to some extent on the speaker's voice characteristics.

IX Insert pauses of a few seconds between recordings.

X Keep a written record of the recorded speech with the tape.

Note: Many cassette and open-reel recorders are sensitive to stray magnetic fields of other electronic equipment, such as switched-mode power supplies, tv sets, CRT's, etc. Therefore do not place the recorder too close to such a device.

7. GETTING STARTED

Make sure that you have the required system configuration: At least 400 kbyte of RAM, the Tecmar or Scientific Solutions IEEE488 card and the Hercules graphics card.
Check the SAB cable and switch on the SAB power.

To start up the speech software insert the OM8210 diskette in drive A and switch on the system power or perform a system reset with [Ctrl]+[Alt]+[Del].
At power-up the system will first test the memory before DOS is booted.

The AUTOEXEC.BAT file will be processed automatically to initialize the system for the speech program:

```
- ECHO OFF              ; switch off system messages
- CLS                   ; clear screen
- ECHO starting up OM8210 speech processing software...
- DOSTIME-2             ; copy battery clock to system clock
                          (Tecmar Captain Card only)
- ECHO.
- DATE                  ; enter system date if required
- ECHO.
- TIME                  ; enter system time if required
- MODE COM1:9600,N,8,1  ; set mode of serial port 1
- MODE COM2:9600,N,8    ; set mode of serial port 2
- HGC FULL              ; set mode of Hercules graphics card
- PATH A:\              ; set command search path
- VERIFY ON             ; set disc write verify
- BREAK ON              ; enable break
- ECHO.
- ECHO Loading speech program...
- SP2                   ; load speech program
- ECHO ON               ; switch on system messages
```

When the system memory is at least 400 kbyte the program will be loaded and starts with a configuration test.
The following text is displayed on the screen (error messages in parentheses):

```
Checking processor type and speed (do NOT interrupt !)...
Testing hardware configuration:
IEEE488/IEC625 card...
   (CHECK TECMAR IEEE488/IEC625 card !!)
Graphics card...
   (CHECK HERCULES GRAPHICS CARD !!)
Memory size... xxxxxx bytes = yy frms = zz sec.
   (not enough, add memory)
Testing interface to the Speech Adapter Box...
   (SAB XXX ERROR nnn)
   (CHECK SAB IEC CABLE AND POWER)
   (Retry or Quit?)
```

At this point the program switches from "text" mode to "graphics" mode. All program messages will appear on the "graphics" screen. Any DOS messages will be displayed on the "text" screen which is now invisible.

Next the analysis tables are initialized.

Now the main command level is entered from which you can start any of the following SPEECH modes:

Analysis Code Eprom File Option Parameter Sample DOS.

The modes are invoked by typing the initial uppercase letter of the mode name. Using a lowercase letter will cause the message "Please press [Caps Lock] key !!" to be displayed.

Pressing [D] quits the speech program and returns to the DOS command level where you can use the DOS internal commands (e.g. DIR, COPY) and the external commands FORMAT and DISKCOPY.

Note: The speech program is aborted when you return to DOS!
Make sure that all valuable data are stored on disk.

7.1 Error messages

Three kinds of errors may occur when running the speech program: SAB errors, Pascal errors and DOS errors.
SAB errors indicate a malfunctioning of the IEEE488 communication. A message is displayed together with a number identifying the driver routine in which the error occurred. To recover from such an error it is sometimes necessary to switch off the SAB unit and turn it back on again. Going to the main command level and pressing [Q] will then re-initialize the hardware.

In case of a Pascal error a message is displayed identifying the program section in which the error occurred, followed by an error number. A list of the most common Pascal errors is included in Appendix A.

As mentioned before a DOS error is not visible on the "graphics" screen which is used by the program.
When a DOS error should cause the speech program to halt inadvertently it is possible to return from "graphics" to "text" mode in the following way:
- Press [A] to abort the speech program
- Type "SC"[Enter] to switch from "graphics" to "text" mode.

The DOS error message will now become visible.

8. KEYBOARD AND COMMANDS

All commands are single-character commands: one keystroke activates the command.
Most commands are invoked using uppercase letters. It is therefore advised to press the [Caps Lock] key at the start of the program.

Some commands require the use of the [Shift] or the [Ctrl] (control) key, e.g. in the Parameter mode [A] provides audio output of "edit" parameters, [Shift]+[A] of the "input" parameters and [Ctrl]+[A] of the original "samples".

All destructive commands - the ones that can overwrite or destroy valuable data - are protected by a "Y/N" confirmation.
Pressing [Y] or [Shift]+[Y] confirms command execution. Any other key will prevent the command from being executed.

The grey [+] and [-] keys are used for general volume and level control as well as increment/decrement actions in the Parameter mode.
For increment/decrement over a range of frames the [Ins] and [Del] keys are used. Range operations require confirmation with [Y] or [Shift]+[Y] to be executed.

In the Parameter mode the cursor left/right keys on the numeric keypad are used for cursor movement. Pressing the [Ctrl] key together with one of these keys provides fast cursor movement.
With the cursor up/down keys a parameter can be selected.
In the Code and File modes the cursor left/right keys can be used to move through the text when entering or modifying a description.

In the Parameter mode the FUNCTION keys [F1] to [F10] provide audio output of 10 to 100 frames starting from the cursor. Simultaneous use of the [Shift] key will play the frames preceeding the cursor.

In the Parameter mode the available commands can be displayed all at once using the H(elp) command or line by line by pressing [?] or [/].

The [Esc] (escape) key can be used to halt the analysis or EPROM writing. If you stop the analysis, the input parameters are valid up to the frame where the analysis was aborted.
An EPROM contains validated data up to the address being programmed at the time [Esc] was pressed.
The [Num Lock] and [Scroll Lock] keys must not be used in the speech program. If you toggled one of them the cursor control in the parameter editor may not work properly.

Use the grey [<—] (backspace) key to rub out preceeding keystrokes.

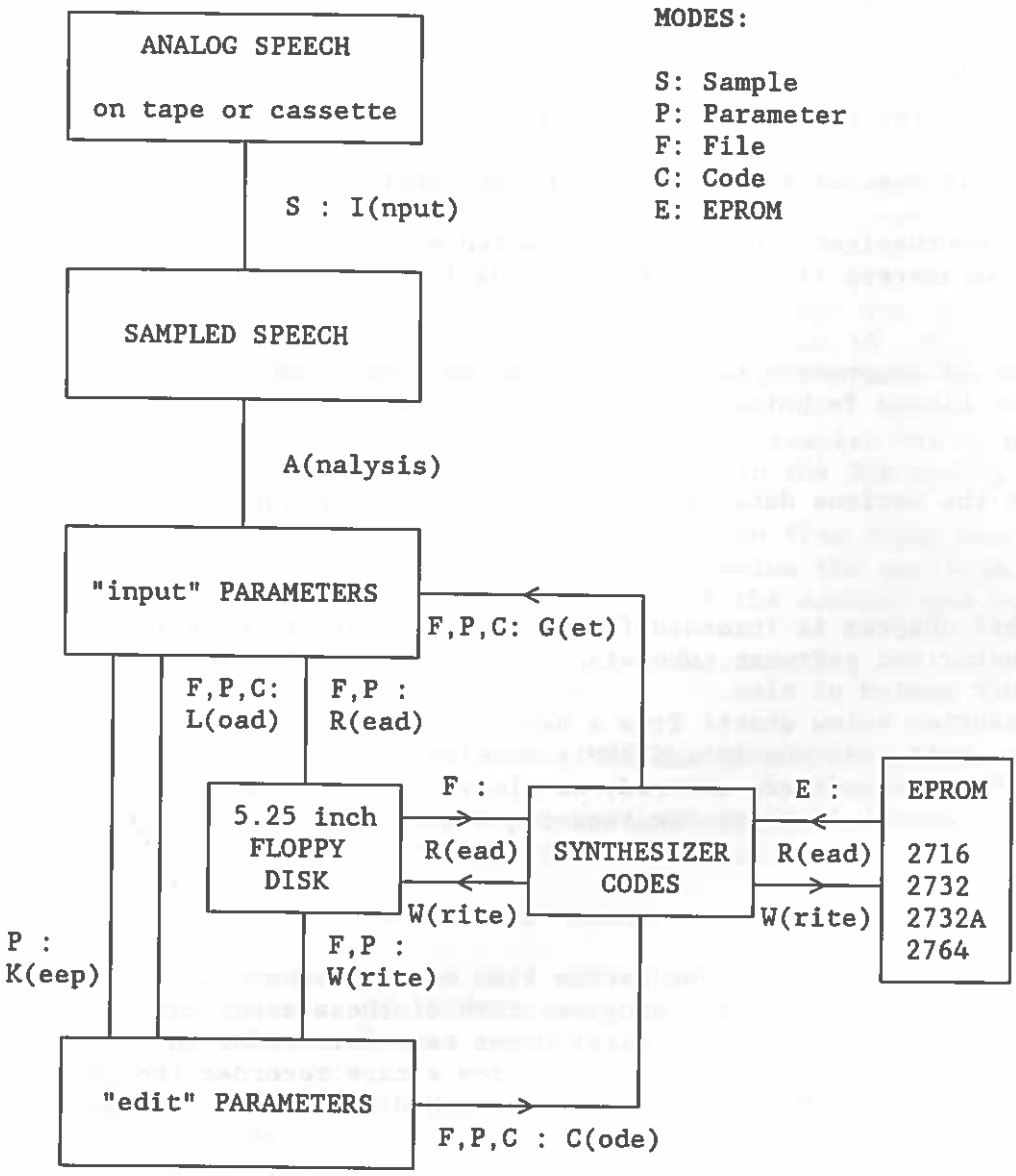

Figure 2. Speech coding data flow.

9. SPEECH CODING DATA FLOW

9.1 Introduction

The system knows three types of speech data:

1. Samples: 12-bit digital representation of the analog input signal.

2. Parameters: synthesizer control data obtained with the analysis program, one set of parameters (1 frame) representing 12.8 ms of speech in default setting.

3. Codes: a set of compressed parameter codes for direct use by the PCF8200 (format: see Elcoma Technical Publication 217) which also includes an address map.

Figure 2 shows the various data flow paths that may occur during speech processing.

The rest of this chapter is intended for those users not yet familiar with the Speech Adapter Box software, who wish to see some results in a relatively short period of time.
The tutorial section below starts from a newly booted system and goes on to input, analyse, edit and code into EPROM a section of speech.
Not all available commands are covered, so please refer to the later sections of this manual if there are some other actions you wish to perform.

9.2 Tutorial

The main menu is on the screen, the bottom line of which shows us all of the main modes available in the SPEECH program. Each of these modes can be called by a single key press of the first upper case letter for the command.
Our objective is to take an analogue signal from a tape recorder (to get the signal on tape follow the Ten Commandments described in Ch.6 of this manual) and produce a coded EPROM.
Key entries are described using the syntax presented in Ch.1 Introduction.
In the following sections of this tutorial it is assumed that "Caps Lock" mode has been activated.

INPUT

[S] selects the Sample mode - this is used to digitize the input sound and to select the portion of the sound we require.

[I] is used to Input the speech. The computer tells you the maximum amount of input possible (in seconds). This is dependent on the size of the available memory. If you require less than the maximum, enter the number of seconds required.

[Enter] Confirms the maximum or modified input time. Start the tape recorder just prior to the required expression.

[Y] Starts the computer sampling of the input sound. When completed the computer 'beeps' and after a short pause shows a screen of signal amplitude (vertical) versus time (horizontal). Stop tape recorder.

It should be relatively easy to identify your expression on the screen. If the amplitude is not near the top of the graph we must increase the ADC level by [L][+]...[+]. If flat tops show in the amplitude curve we must decrease the ADC level by [L][-]....[-].

The sequence below is repeated until either 'Maxamp' (on the second header row) is greater than 1800 or 'Clips/s' is less than 10 ('Maxamp' is replaced with 'Clips/s' only when amplitudes greater than 2047 are found).

[A] Allows us to hear the digitized samples via the Digital-to-Analogue converter in the SAB system.

We can select a particular phrase or expression from input sound by adjusting the 'BEGIN' and 'END' pointers. Examine the amplitude waveform on the screen and estimate the start and end of the section you require.

[B] Selects the BEGIN pointer and defaults to 1. Type in the new value.

[Enter] Loads the new value into the second header row.

[E] Selects the END pointer and defaults to "end of samples". Type in the new value.

[Enter] Loads the new 'END' value.

[A] Allows you to hear the selected expression. Repeat the BEGIN and END exercise until you have achieved the desired speech segment.

[Q] Quits the sample mode and returns to the main menu.

ANALYSIS

[A] Enters the Analysis mode of operation. The system indicates the number of frames to be analysed (from 'BEGIN' to 'END') and asks if the voice to be analysed is Male or Female.

[M] or [F] to select the correct voice table (children generally give better results on Female tables).

[Y] Confirms the action and starts analysis. A frame counter can be seen indicating the number of frames completed.

The computer is now extracting all of the speech parameters (amplitude/pitch/voiced-unvoiced/formant frequencies/formant bandwidths) from the input sound. When all frames are analysed the computer immediately returns to the main menu screen.

PARAMETER EDIT

[P] Selects the Parameter mode. The computer will recognize that we have done some analysis and prompt us to load the resulting parameters into the 'edit' memory.

Starting at the bottom of the screen the graphical representations are of:

Bottom row - Representation of each frame; when we come to enter frame duration editing, changes will occur here.

Row 2 - Representation of amplitude for each frame.

Row 3 - Representation of pitch for each frame, where a vertical bar indicates a frame which is unvoiced.

Row 4..7 - The horizontal bars represent the formant and maybe 8 frequencies while the vertical bars show the formant bandwidths. Working with male speech all 5 formants are present, for female speech 4 exist. Formant 1 is at the bottom.

We recommend you now save the parameters as they were produced by the analysis.
For the following commands it is assumed that a speech file directory (FDIR) is present on the specified filer pathname. If not please refer to chapters 4 and 15.

[F] Selects the Function menu. Actions in this area affect the whole file from 'BEGIN' to 'END'.

[W] Writefile - prompts with the next available free file number.

[Enter] Accepts that file number and presents us with a 58 character string to describe the file. Conventions for this string are not rigid but usually we end up with a structure which looks similar to this:

R3F PROJECT - MINIMUM FEE

Where "R3F" means Rough data, quality level 3, Female voice. The alternatives to Rough are Polished or Frameduration-edited, while quality ranges from 1 (poor) to 5 (excellent) and the alternative to Female being Male or Child.

[Enter] To terminate the 58 character string. The total file description is shown.

[Y] Confirms you are satisfied with it and begins writing the data to disc.

When using our new analysis method very little post-analysis editing is required. However, for some voices it may be necessary to check the pitch extraction values and check the voiced/unvoiced decisions.

Pitch errors

[cursor up/down] Selects which parameter is currently being worked with. The parameter selected can be seen as the second item in the second header row. The first item in that row being the current frame at the cursor.

Select the 'PT' (pitch) to be the current parameter as described above or if you have been following these instructions you will already be there. The memory of the computer has been divided into four main parts:

1) 'Samples' memory containing the digitized input sound.
2) 'Input' memory which is the location of the parameters after analysis and used as a back-up memory.
3) 'Edit' memory - where we loaded the parameters to appear on screen.
4) 'Code' memory - the location of PCF8200 codes once they have been created.

[A] Outputs the 'edit' memory via the speech synthesizer.

[Shift]+[A] Outputs the 'input' memory via the speech synthesizer.

[Ctrl]+[A] Outputs the 'Samples' memory via the SAB's DA-converter to reproduce the original speech.

To return to 'PT' (pitch) editing. Most pitch errors are octave errors (factor of 2) and are big jumps in sound. This makes them readily identifiable by

[O] Continuous sound output for the frame at cursor - move the cursor through the screen listening for discontinuities.

Otherwise a visual scan of the pitch contour (row 3) will probably show the error. If the error is not on the current screen:

[N]	Selects the Next screen full of data.
[P]	Loads the Previous screen of data. The first and last pages of data are indicated below the header rows.
[D]	Displays one page forward from the cursor position - useful if the change is on the edge of a page.
[cursor left/right]	Using these keys move the cursor to the left of the start start of the identified errors. The cursor frame can be highlighted by pressing [space].
[S]	If the error is a single frame this command will Smooth the transition between the previous and next frames.

If more than a single frame is in error:

[W]	Sets a Workpointer at the current cursor position (and updates the fourth item in the second header row).
[cursor left/right]	Move the cursor to the left of the last frame which contains the errors.
[*]	Increases the pitch by an octave (= multiply by 2).
[Enter]	Accepts the default multiplication factor.
[Y]	Confirms the action.
[:]	Decreases the pitch by an octave (= divide by 2).
[Enter]	Accepts the default division factor.
[Y]	Confirms.

Usually this will give a good quality result and a smooth contour. However, if this is not the case set up the Workpointer as before and the cursor to the right of the last error frame.

[I]	Will Interpolate pitch between the two points.
[Y]	Confirms.

For all actions it is recommended that regular monitoring of the audio output takes place to compare the original with the synthesised expression using the [A], [SHIFT]+[A] and [CTRL]+[A] commands. If the resulting speech quality has not improved we can

[R]	Reload the pitch parameter between the workpointer and cursor. This command copies the selected parameter from the 'input' memory into the 'edit' memory.
[Y]	Confirms.

Voicing errors

The other error which may occur with the analysis system is the selection of the voiced/unvoiced switch. This can be seen in the 3rd row up of the display. Those pitch markers which have vertical lines on them are the unvoiced sections of speech. Mostly, the only markers which need checking are those on either side of a voicing change. The easiest way of checking is to select

[O] Mode to give a continuous output of the frame at cursor position.

Select the voiced/unvoiced 'VU' parameter in the second item of the second header row by pressing [cursor up]. Move the cursor across a voicing transition and check for possible errors.
Unvoiced frames having a high B1 value (long vertical marks through first formant, 4th row from bottom) must be considered suspicious. They may have been classified wrongly as being unvoiced.
Then experiment by converting the voicing at a suspicious frame using [-] to make it voiced, or [+] to make it unvoiced - listening all the time to check for a reasonable sound output. If in doubt leave the frame as it was.

We now recommend that the file is again saved (marking it 'P' for Polished) using the following input sequence:

[F][W][Enter](file description)[Enter][Y]

Frame duration

The final act of editing the speech is to reduce the bit rate. To do this we use the feature within the synthesiser for interpolating between the last time frame and the current one. If the characteristics of the speech have few major changes in them, we can increase the frame duration to be twice, three times or five times its original length. Therefore we can save one, two or four frames of data and get our resulting speech into a smaller EPROM.

There are two ways to do FD editing. The hard way we will ignore in this tutorial. The easy way is to use the Auto_fd option from the Function menu, which is an automatic procedure to reduce the number of parameter frames while preserving all important features of the speech.

[F] Selects the Function menu.

[A] Gives access to the Auto-fd.

[Enter] Accepts the default "frame cost", a parameter used to balance speech quality and number of frames.

A frame cost of 3000 will give a high-quality result. You can experiment with frame cost, higher values giving lower bit rates at the cost of lower speech quality.

[Y] Confirms the action to begin.

The system then calculates the frame durations for you and when finished, will display the bit rate for that speech segment.

Save the file on disc (marking it 'F' for FD-edited).

[F][W][Enter](file description)[Enter][Y].

CODING

If we are building up EPROM code as we proceed then the next step is to generate the PCF8200 code for the speech.

[C] Codes the speech and gives an audio output, followed by 20 characters to describe it in the code map.

[Enter] Terminates the description.

[Y] Adds the code to the map in the system.

At the end of the session when the code map contains all the files you require we recommend you save this on disc for either adding to or blowing into EPROM later.

[Q] Quit from the Parameter mode.

[F] Select the file mode.

[T] To change the type of file to "CODES". This can be checked as it is the first item in second row of the header.

[W] Writes file to disc prompting with the first available file number followed by up to 58 characters.

[Enter] To terminate the line.

[Y] Confirms writing the code map and the PCF8200 codes to disc.

This code can be retrieved using the Read command. Assuming we are still in the Filer and the file type is still "CODES":

[R] Reads a file from disc. The system then prompts for a file number. Enter the file number you require followed by

[Enter] The system responds with the file information and asks for a confirmation.

[Y] If file information correct to start readingn code file from disc.

EPROM PROGRAMMING

[Q] Quit the filer.

[E] Enter the EPROM mode.

[1..4] To select the EPROM type. Press the number for the relevant type and load EPROM into socket on OM8210.

[W] to Write the EPROM. Data originate from the "code" memory and consist of an address map followed by the corresponding speech codes.

The system then asks us to give it some addresses:

[Enter] RAM start address - this is nearly always 0.

[Enter] EPROM start address - this may be some other address e.g. if a controlling programme is in front of the speech code. However, if the EPROM is to contain only speech code this is also 0.

[Enter] Accepts the EPROM end address.

[Y] A final confirmation is required.

The system now runs a bit check on the EPROM then begins to write the data. An indication of the number of seconds remaining is given on the screen. When the writing of the EPROM is complete the system automatically runs the Verify program and highlights any errors which exist. The EPROM may now be taken to the target system and installed.

10. MAIN COMMAND LEVEL

In the main command level a menu of SPEECH modes is displayed:

Analysis Code Eprom File Option Prameter Sample DOS.

These modes are described below in the order that you will normally use them (see also figure 3).

Sample: Permits speech input from tape, checking the level and setting BEGIN and END of a speech segment.

Analysis: Extracts formants and pitch from the speech segment between BEGIN and END.

Parameter: Permits editing of the analysis results, reducing the bit rate by using a variable frame duration and generating compressed speech code.

Code: Permits (re)ordering the map of coded messages for EPROM storage.

File: Is used to make disc backups of parameters or compressed codes.

EPROM: Is a built-in programmer to store speech codes into a 2716..2764 EPROM for use in your application.

DOS: Returns control to DOS where you can use the DOS internal commands, prepare new disks and make backups.

Option: Provides extra facilities, e.g. setting default values, changing the system date, switching from IBM to Philips P3100 computer, etc.

Note: When you return to DOS any recorded speech samples, the parameters and the codes are lost because the speech program is aborted. Be sure that you have made disc backups using the File mode before you go to DOS !

All SPEECH modes are accessed by typing the initial uppercase letter of the mode name. You can leave any SPEECH mode using Q(uit).

Input level and output volume are controlled by the [+] and [-] keys.
Level control is invoked by pressing [L].
Volume control is invoked by pressing [V] for synthesizer volume or [Shift]+[V] for DAC volume.
After selecting a potmeter ([V], [Shift]+[V] or [L]) you can change its setting with [+] or [-]. Pressing any other key will exit potmeter control.

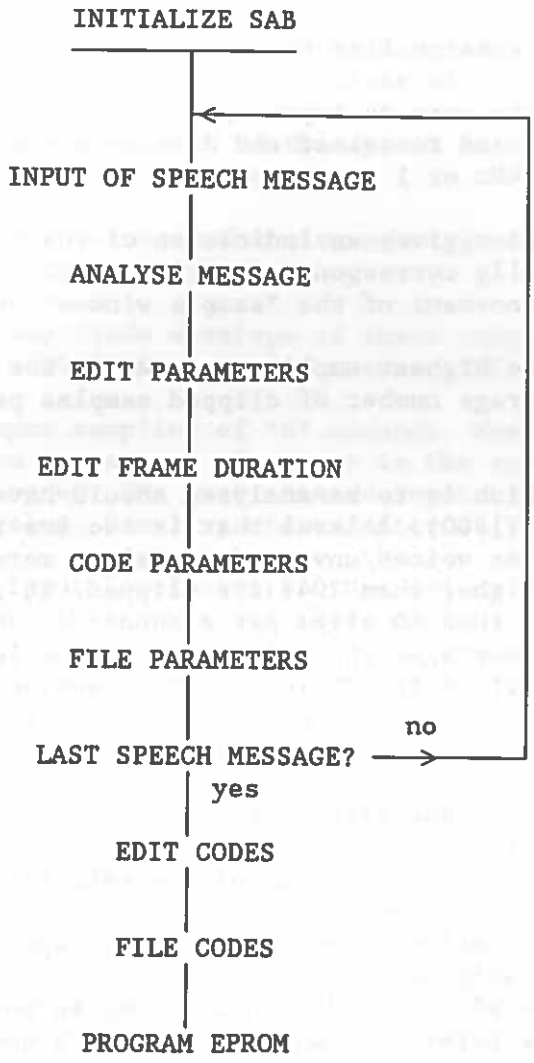

Figure 3. Speech coding procedure

11. SAMPLE mode

The Sample mode is entered from the main command level using the [S] command.
This mode enables the user to input speech from tape into the system memory, to check the level, and to select and display a speech segment. The default sampling rate is 10kHz or 1 sample per 100 μs.

The display of samples gives an indication of the maximum amplitude found in each "frame", normally corresponding to 128 samples or 12.8 ms. A "frame" represents the advancement of the "sample window" during analysis.

The header shows the highest amplitude found in the recorded utterance (Maxamp) or the average number of clipped samples per second of speech (Clips/s).

A speech segment which is to be analysed should have a sufficiently high amplitude (Maxamp > 1800): A level that is too low reduces the signal-to-noise ratio and makes voiced/unvoiced decisions more difficult as well. Signal amplitudes higher than 2047 are clipped. It is advised not to have a a higher clip count than 10 clips per second.
Note that for long messages the clip count can be low while the level is too high for a small part of it. Therefore it is advised to also check clipping by ear: Audibly distorted parts of the speech always have a clip count which is too high regardless of the value displayed.

In this software the frame size (referred to as "speed") can be specified by the user, both for the analysis and for synthesis.
The smaller a frame the more detail of the original message is preserved at the cost of a higher bit rate.
When synthesis speed differs from the analysis speed the speech rate will be different from the original recording.
Standard speeds are 88, 104, 128 and 176 samples per frame. Non-standard analysis speeds are permitted, whereas synthesis speeds are limited to the standard values.
For a more extensive explanation of the analysis procedure please refer to Chapter 12.

This software release (version 1.x) includes the possibility to store and retrieve "samples" by means of a SAMPLES disc file.
Files of this type have filenames composed of a letter 'S' followed by a 3-digit number (e.g. S008) and are stored using the default "pathname".Commands to be used are W(rite) and R(ead).
Writing to an existing file will result in overwriting the old file. Sample files can only be deleted via DOS using the "DEL" (or "ERASE") command.
SAMPLES files can be accessed only in the Sample mode and are primarily intended for use in the Batch mode of operation (see section 17.2).

11.1 SAMPLE mode commands

[A]	gives Audio output of the input samples between BEGIN and END via the 8-bit DA-converter of the SAB.
[B]n	sets the BEGIN pointer of the speech segment to frame "n" (default: 1).
[E]n	sets the END pointer of the speech segment to frame "n" (default: end of samples).
[D]n	Displays amplitude envelope of input samples from frame "n" (default: BEGIN) to end of screen or END.
[I]n	starts Input sampling of "n" seconds. Maximum input time depends on the amount of memory in the system and on the selected speed. The last entered input time is kept as default value. Sampling starts after a [Y] confirmation.
[L]	accesses input Level control. Use [+] or [-] to change the level. Any other key exits level control.
[P]n	Plots the sampled waveform starting at frame "n".
[R]n	Reads SAMPLES file "n" from the default "pathname" on disk into "samples" memory. BEGIN and END pointers are adjusted. Confirm with [Y] to execute.
[S]n	sets analysis Speed (window advancement) to "n" samples per frame (default: 128 = 12.8 ms). Resynthesis speed will automatically be adjusted to the nearest available value (i.e. 88, 104, 128, or 176) in order to preserve the original speech rate as much as possible. If a non-standard analysis speed is chosen, a warning will be issued.
[Q]	Quit: exits the Sample mode.
[V]	Selects DAC volume control. Press [+] or [-] to change the volume. Any other key exits volume control.
[Shift]+[V]	same for synthesizer volume.
[W]n	Writes the contents of the "samples" memory from BEGIN to END to SAMPLES file "n" on disk under the default "pathname". Overwrites an existing file of the same number. Confirm with [Y] to execute.
[+] or [-]	controls DAC output volume. Any other key exits volume control.

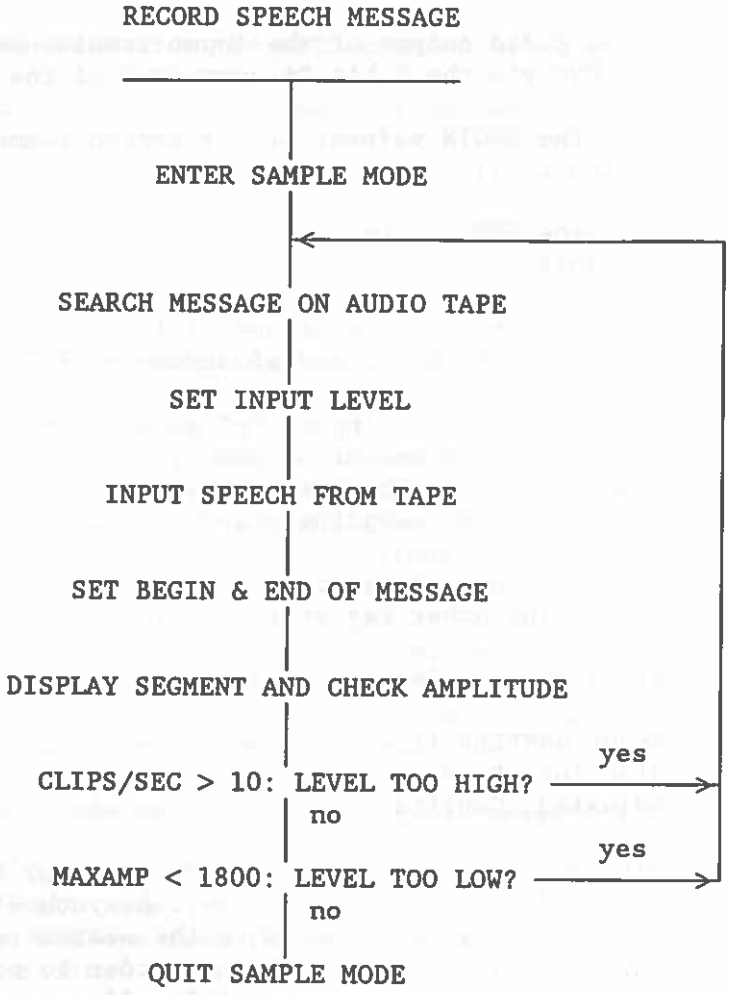

Figure 4. Speech input procedure

12. ANALYSIS mode

The Analysis mode is accessed from the main command level by means of the [A] command.
The Analysis will perform formant and pitch extraction on the speech segment marked by the BEGIN and END pointers in the Sample mode.
A sample window is passed over the message in steps equal to the analysis speed. For each window position a set of speech parameters is determined. This set of parameters is referred to as a "frame".
In the synthesis each "frame" is used to generate output speech for a duration determined by the synthesis speed.

Since male and female voices differ considerably in frequency characteristics, the analysis has two quantization tables, one for each type of voice. For male voices an analysis in five formants is performed while for female voices only four formants are determined (see: Elcoma Technical Publication 217).
Analysis can be invoked after sample input and segmentation. The screen will show the number of frames to be analysed as well as the BEGIN and END pointer values from the Sample mode. Next the user must select Male or Female voice analysis. Analysis is not started until after a [Y] confirmation. During analysis the display shows the type of voice selected, and a counter indicating the number of frames analysed sofar.
When in the Sample mode the analysis speed was set to a value other than default (128), this will be displayed before the confirmation request.

Analysis requires about 180 seconds per second of speech (standard IBM XT). Low-level parts, such as pauses, are not completely analysed. Therefore the analysis of entire expressions may be faster.

Analysis will only halt when completed or when the [ESC] key has been pressed, followed by a [Y] confirmation. In the latter case the parameter input memory will contain valid parameters up to the frame where the analysis was aborted.
After completion of the analysis the program automatically returns to the main command level.

When you have selected "Spectrum display" in the Option mode you will see the graphic representation of the Fourier power spectrum of each frame. This option will slow down the analysis.

The analysis results are stored in the parameter "input" memory. They can be manipulated in the Parameter mode. Before this can be done, the analysis results have to be copied from the "input" into the "edit" memory using the L(oad) command.

13. PARAMETER mode

The Parameter mode is entered from the main command level by means of the [P] command.
In this mode the parameters obtained from the speech samples with the analysis program, can be optimized in quality and bit rate using the Parameter editor commands.

Two parameter memories are available:

1. The "input" parameter memory is used by the analysis program to store its results. It is also used by the G(et) command that retrieves parameters from coded versions and by the R(ead) command in File or Parameter mode when reading a parameter file from disc.

2. The "edit" parameter memory contains the parameters you can manipulate directly and which are graphically displayed on screen. All parameter data "out" streams originate from the "edit" memory.

The "input" parameters can also be used as a backup for the "edit" parameters by using the K(eep) command. The L(oad), R(eload) and T(ail) commands can be used to transfer "input" parameters to the "edit" memory. The user can switch between "edit" and "input" memory by means of the (e)X(change) command.

13.1 Parameter display format

In the Parameter mode up to 180 frames can be displayed simultaneously as shown in figure 5 for 5 frames. The selected frame is preceeded by the cursor line and displays all parameters from FS at the bottom to F5/B5 (F4/B4 for female voices) at the top of the screen.

Synthesis speed (FS) is displayed as short vertical lines below the baseline. For FS=88 no lines are shown in the display. One line is displayed for FS=104, two for FS=128, and three for FS=176. The code value (shown in the second line of the header) equals the number of samples per standard synthesis frame. When FS is decreased the speech rate will increase and vice versa.

The frame duration (FD) indicates the frame update period in standard frames, the length of which is determined by FS. The FD parameter can be used to reduce the speech bitrate (see section 13.4). It is displayed directly above the baseline as a vertical tick mark, the size of which is a measure of the FD value (0 = 1 st.frame, 1 = 2 st.frames, 2 = 3 st.frames, 3 = 5 st.frames). FD=4 indicates a "skip" frame (no tick mark at all). A skip frame is ignored during coding.

For a standard FS setting of 128 samples per frame the FD parameter gives the following frame lengths: 0 = 12.8 ms, 1 = 25.6 ms, 2 = 38.4 ms, 3 = 64 ms.
Above we find the amplitude (AM) display bar. Its vertical position is a measure for the AM code value (0-15). The AM code is logarithmically related to the excitation source amplitude of the synthesizer chip.

The pitch (PT) is the frequency in Hz of the voiced source signal. Its value can range between 0 and 511 Hz. The vertical position of the pitch display bar is logarithmically related to the pitch value. Pitch values of 50 Hz and lower are displayed at the same bottom level.

The absence of voicing (VU=1) is indicated by a vertical bar through the PT bar. For these frames a noise source is used in synthesis.

The formant frequencies (F1, F2, F3, F4, and - for male voices only - F5) are displayed on a linear vertical scale, each with its own offset to avoid overlapping display curves.
Formant frequency increases with increasing code value. See Appendix B for a list of formant frequencies versus code values.
Formants are displayed above the pitch curve in the order of ascending frequency (F1, F2, F3, F4, F5).
F1 and F2 can have 32 values (0..31), F3 and F4 can have 8 values (0..7), F5 can have two values (0-1).

Formant bandwidths (B1, B2, B3, B4, and - for male voices only - B5) are displayed as a vertical bar through the corresponding formant frequency bar. The length of the bar is a measure of the inverse of the bandwidth code. In this way a small bandwidth (high filter-Q or high amplification) will be displayed with a long vertical bar.
B1 and B2 can have 8 values (0..7). B3, B4, and B5 can have 4 values (0..3). See appendix B for a list of bandwidths versus code values.

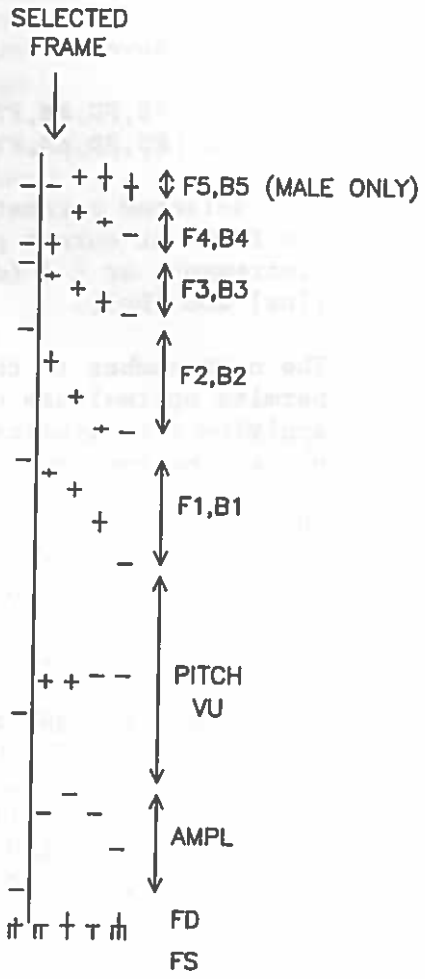

Fig. 5. Parameter display

13.2 Pointers in Parameter mode

The parameter "edit" pointers are displayed on the second line of the screen heading together with the selected parameter and the synthesizer DAC control code.

The first number on this line is the cursor position. The cursor is controlled by the [cursor left] and [cursor right] keys. Using the [Ctrl] key simultaneously provides "fast" control.

The selected parameter can be changed with the [cursor up] and [cursor down] keys.
It can be moved through the following sequence:

male : FS,FD,AM,PT,VU,F1,B1,F2,B2,F3,B3,F4,B4,F5,B5.
female : FS,FD,AM,PT,VU,F1,B1,F2,B2,F3,B3,F4,B4.

The "selected parameter" indication is followed by its actual code value for the frame at cursor position. This value can be changed using e.g. the [+] (increment) or [-] (decrement) keys or the corresponding range operations [Ins] and [Del].

The next number is the synthesizer DAC control code (DA). This control code permits optimal use of the range of the 11-bits wide synthesizer DAC, by applying a multiplication factor to the signal before DA-conversion takes place. The code values range from -3 (a factor of .25) to 10 (a factor of 3.5).
The DAC control code can be accessed by keying [Ctrl]+[D]. It can be changed with the [+] and [-] keys. Pressing any other key exits the DAC control mode. The DAC control code is valid for all frames between Begin and End.

Four other frame pointers are shown:

- The WORKPoinTeR is used in commands that can operate on a range of frames. When the WORKPTR is used in a command it will be displayed as a second cursor line to indicate beginning and end of the range operation. Such commands will request a [Y] confirmation.
- The BEGIN and END pointers indicate where a message starts and ends for audio output, code generation, and storage on disk in a parameter file.
- The EOF (end of file) indicator shows the position of the last valid frame in the "edit" memory.

The WORKPTR, BEGIN, and END pointers can be set by placing the cursor on the required position and then pressing [W], [B], or [E] respectively.

When "O" is displayed at the end of the second line the synthesizer is giving continuous output of the frame at cursor position. Output may be switched on or off with [O].

13.3 Parameter editor commands

There are many commands available in this mode. Normally you will only see the first command line. The other lines can be displayed sequentially by pressing [?] or [/]. All lines together can be shown with the H(elp) command. Pressing any other key returns the display of the first command line.

Most commands are activated by typing a single uppercase character as shown in the "Help" display and the command list below.

The Parameter editor commands can be subdivided into seven groups:

1. Parameter transfer: Keep, Load, Reload, Tail, eXchange.

2. Pointer handling : cursor: [cursor left/right], [<] or [,], [>] or [.]
   selected parameter: [cursor up/down]
   range: Workpointer, Begin, End.

3. Display : Display, [space], Previous page or
   [Pg Dn], Next page or [Pg Up]
   [Ctrl]+[P] or [Home]: first page
   [Ctrl]+[N] or [End]: last page

4. Change parameter : Smooth, Interpolate, [+]/[Ins] increment, [-]/[Del] decrement, [*] multiply pitch, [:] divide pitch.

5. Audio output : Audio, Output (1 frame continuously),
   [F1]..[F10] (output 10 .. 100 frames)
   Z (code output).

6. Code handling : Code generate, Get code.

7. Miscellaneous : Function, Help, Make, Unmake, Volume/Level control,
   Read and Write files,
   [Ctrl]+[D] synthesizer DAC control.

Detailed description of parameter editor commands:

[A] Audio output of "edit" parameters between BEGIN and END.

[Shift]+[A] Audio output of "input" parameters from BEGIN to END.

[Ctrl]+[A] Audio output of "samples" between BEGIN and END as set in Sample mode.

[B] sets the "edit" parameters BEGIN pointer to cursor position.

[C] generates EPROM Code from the "edit" parameters between BEGIN and END. Before coding the message is output and the system asks for a description of max. 20 characters. After a [Y] confirmation the generated code is stored in the "code" memory.

[D] Displays "edit" parameters starting from the cursor.

[Ctrl]+[D] gives access to the synthesizer DAC control code. To change the setting use the [+] and [-] keys. Any other key exits DAC control. Use this control code to match utterances/words in output level.

[E] sets "edit" parameters END pointer to cursor position.

[F] Function menu: operations on the entire "edit" memory (from BEGIN to END). This OM8210 version (1.1) contains:

[A]	Auto_fd	-	performs automatic setting of frame duration FD
[C]	Clear_fd	-	resets FD to 1 standard frame (FD=0).
[D]	Double_fd	-	sets FD to 2 standard frames (FD=1) (skips every other frame).
[L]	Loadpar	-	reloads the selected parameter from the "input" into the "edit" memory.
[+]	incr.	-	increments the selected parameter.
[-]	decr.	-	decrements the selected parameter.
[P]	Pathname	-	Enter new pathname for File mode operations. Confirm with [Y].
[R]	Readfile	-	reads a parameter file from disc to "input" memory (see File mode).
[W]	Writefile	-	writes a parameter file from "edit" memory to disc (see File mode).

[G]n Gets code number "n" from the "code" memory into the parameter "input" memory. Uncoded frames are replaced with copies of the following coded frame (but with FD=4). Before an [Y] confirmation is requested the code is output via the synthesizer.

[H] Help displays a menu of all available commands.

[I] Interpolates the selected parameter between WORKPTR and cursor. Confirm with [Y] is requested. FD cannot be interpolated. PT (pitch) is interpolated logarithmically.

[K] Keep copies the "edit" memory contents into the "input" memory. This command can be used to save intermediate editing results. Confirm with [Y].
It is recommended to keep the "edit" parameters before starting FD-editing. In this way the codes with and without variable frame duration may be compared.

[L] Load copies the "input" memory contents into the "edit" memory. BEGIN, END and EOF pointers are adjusted. Confirm with [Y].

[M]n Makes "n" copies of the frame at cursor position and inserts them after the cursor. Confirm with [Y]. This command is used to stretch parts of words or make longer pauses between words. BEGIN, END and EOF pointers are recalculated.

[N] or [Pg Up]	displays Next page for editing. No action when the last page is already displayed.
[Shift]+[N] or [End]	displays the last page of the "edit" memory. This facilitates fast resetting of the END pointer.
[O]	toggles continuous Output via the synthesizer of the frame at cursor position. Moving the cursor or going to another page does not affect the switch setting. Any other audio output command will switch off the continuous frame output.
[P] or [Pg Dn]	displays Previous page for editing. No action when the first page is already displayed.
[Shift]+[P] or [Home]	displays first page of the "edit" memory. This facilitates fast resetting of the BEGIN pointer.
[Q]	Quit: exits the Parameter mode.
[R]	Reload restores the selected parameter between WORKPTR and cursor from the "input" memory. Confirm with [Y]. When "input" and "edit" memory lengths do not match a warning is issued.
[Shift]+[R]	Reloads entire frames between WORKPTR and cursor. Otherwise the same as Reload.
[S]	Smooth interpolates the selected parameter of the frame at cursor position between its neighbouring frames. FD cannot be smoothed. PT (pitch) is smoothed logarithmically.
[T]	Tail puts the "input" memory contents at the end of the "edit" memory and recalculates EOF and END pointers accordingly. Confirm with [Y]. This command can be used to concatenate a longer message from existing parameter files, containing words or parts of a sentence.
[U]n	Unmake deletes "n" frames from the frame at cursor position onwards, BEGIN, END and EOF pointers are recalculated accordingly. Confirm with [Y].
[V]	selects Volume control for DAC output. Press [+] to increment or [-] to decrement. Any other key exits volume control mode.
[Shift]+[V]	selects volume control for synthesizer output. Otherwise the same as [V].
[W]	sets the WORKPTR to the cursor position.
[X]	eXchanges the "input" and "edit" memory contents. Confirm with [Y].
[Z]n	gives output of number "n" from the "code" map.

Special characters:

[F1]..[F10] gives output of 10..100 "edit" frames following the cursor.

[Shift]+ [F1]..[F10] gives output of 10..100 frames preceeding the cursor.

These commands can be used to judge the quality of parts of a message or to locate certain segments within the message.

[+] Increments the selected parameter at the cursor position.

[Ins] Increments the selected parameter from WORKPTR to cursor.

Note: For AM(plitude) no increment over range is performed when the code value is 0. In this way pauses are not lost when increasing the amplitude.

[-] Decrement the selected parameter at the cursor position.

[Del] Decrements the selected parameter from WORKPTR to cursor.

[*]n Multiply the pitch parameter by a factor (n/100) between WORKPTR and cursor. Default factor is 2(=1 octave). When the pitch parameter (PT) has not been selected there is no action. Operation on more than one frame requires a [Y] confirmation.

[:]n Divide the pitch parameter by a factor (n/100) between WORKPTR and cursor. Otherwise the same as Multiply.

[<] or [,] moves the cursor directly to the first frame of the displayed page.

[>] or [.] moves the cursor directly to the last frame of the displayed page.

[space] Pressing the space bar highlights the frame at cursor position by erasing and redisplaying it.

Figure 6 shows the recommended procedure for editing the speech parameters as they are produced in the Analysis mode.

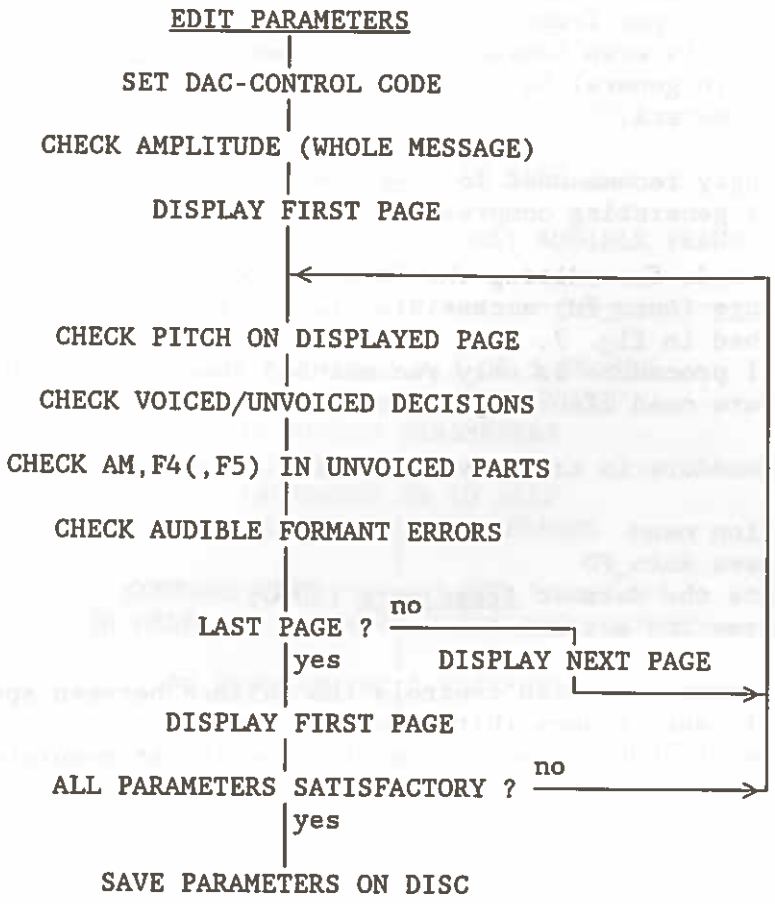

Figure 6. Parameter editing procedure

13.4 Frame duration editing

As explained in Chapter 12 each frame represents the speech parameters corresponding to a standard frame. The duration of a standard frame is determined by the synthesis speed (FS) parameter and determines the maximum data rate for a message.

When a message contains parts not requiring frequent updating of speech parameters (such as vowels and glides) the speech can be reduced in bit rate using the frame duration (FD) parameter. This parameter determines the frame update period in standard frames. When for instance the FD parameter of a frame was set to 1 this frame will be used to synthesize speech lasting 2 standard frames (of duration FS samples).

When the FD parameter is incremented the corresponding number of preceding frames on the screen are "skipped". When the FD parameter is decremented the preceding frames reappear.

This feature can be used because the synthesizer interpolates all speech parameters in 8 steps per frame, regardless of the value of FD.
The pitch parameter is even interpolated 8 times per standard frame.
Therefore, FD can in general be increased if there is a (nearly) linear change in all parameters.

Note: It is strongly recommended to save the final FD-edited parameters on disc before generating compressed code.

There are two methods for editing the FD-parameter of an utterance: an automatic procedure (Auto_fd) accessible via the Function menu, and a manual procedure described in fig. 7.
Use of the manual procedure is only recommended when the results of the automatic procedure need minor adjustments.

The automatic procedure is called with the following sequence:

[F]	Function menu
[A]	accesses Auto_FD
[Enter]	accepts the default frame cost (3000)
[Y]	Confirms the action.

Frame cost is a parameter which controls the balance between speech quality and the number of coded frames (bit rate).
The default value of 3000 gives high quality results at moderate bit rates (1500-2000 bits/s).
Higher values of "frame cost" produce lower bit rates at the cost of lower speech quality.

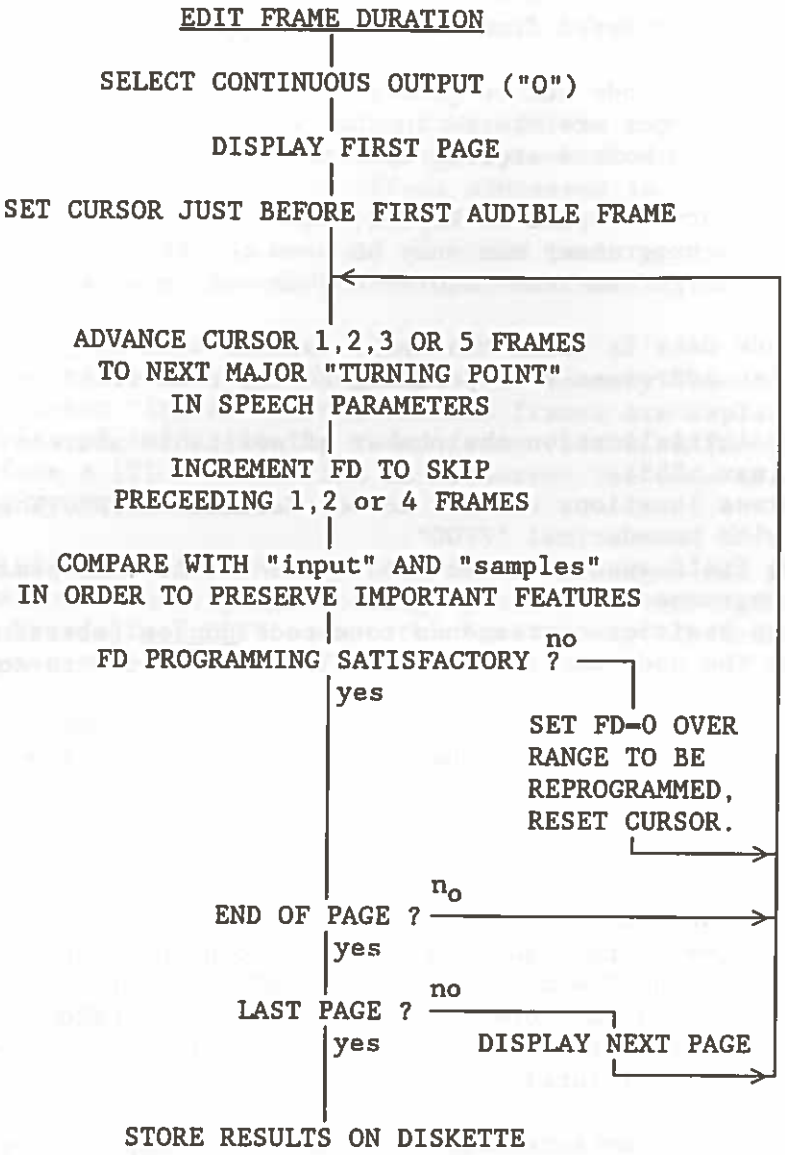

Figure 7. Manual frame duration editing procedure

14. CODE Mode

The Code mode is entered from the main command level by using the [C] command.
Compressed speech code can be generated from the data in the parameter "edit" memory. Codes are stored in the "code" memory. The Code mode enables code manipulation before storing data in EPROM.

The "code" memory occupies 32 kbytes, equal to the size of a 27256 EPROM. SAB's built-in programmer can only be used to program EPROM devices from 16 to 64 kbits. Larger devices require the use of an external programmer.

When the code data is valid the memory starts with an address map: a list of 2-byte offset addresses each pointing to the code field of a different message.
At code map initialization the number of available address locations can be specified (max. 255).
Unused address locations are set to hexadecimal "FFFF". The end of the map is marked with hexadecimal "FF00".
Unused code field space is also set to "FF" so it will remain blank when an EPROM is programmed.
Each address position corresponds to a code number (starting from 0).
At start-up the code map is automatically initialized to contain 16 addresses.

The order of the messages in the map can be changed using the (e)X(change) command. Messages can be removed from "code" memory using the K(ill) command.

It is possible to add speech code to an existing code map in an EPROM provided the map still has unused entries and there is sufficient space for the speech codes. First you read the EPROM contents (see Chapter 15: EPROM mode) into the "code" memory. The empty code space is filled with "FF" so that it can be used to store any new messages. The EPROM contents can thus be extended provided the code map was not modified by E(xtend), K(ill), (e)X(change) or I(nitiate).

The "code" memory contents together with a code map listing can be saved on disc by using the Write command in the File mode. Before executing the Write command make sure that the file Type has been set to "CODES".
Code files are stored in Intellec-Hex format and can be downloaded to an external EPROM programmer using the "COPY" command from DOS. The default settings of the COM-ports can be modified using the "MODE" command from DOS.

14.1 Code editor commands

[A]n	gives Audio output of code "n" via the synthesizer.
[B]n m	gives output of a Bunch of code messages from number "n" to number "m", each requiring confirmation with [Y]. Output sequencing stops at any other key. When "n" is greater than "m" sequencing wraps around to number 0 when end-of-map is reached.

[C] generates compressed Code from the parameter "edit" memory between BEGIN and END. Following audio output a description can be entered of max. 20 characters. Coding is done after a [Y] confirmation.

[E]n Extends an already existing code address map by "n" entries. After confirmation with [Y] the speech codes are shifted to make room for the extra offset addresses in the map.

Note: When map and codes were loaded from an EPROM it is not possible to write the extended map into that same EPROM without first erasing the device.

[G]n Gets code number "n" from the "code" memory into the parameter "input" memory. Uncoded frames are replaced with copies of the following coded frame (but with FD=4). Before a [Y] confirmation is requested the code is output via the synthesizer.

[I]n Initializes the code map to hold "n" messages and prepares the "code" memory for storing speech code. All bytes are set to hexadecimal "FF" except the "end of map" indicator, which is set to "FF00".

[K]n Kill removes speech code "n" from the address map and crunches the speech codes so that the killed data are lost.

[L] Load copies the "input" memory contents into the "edit" memory. BEGIN, END and EOF pointers are adjusted. Confirm with [Y].

[M] displays the contents of the code Map. Each code number is followed by its offset address, the code length and bitrate (bits/s), and a 20-character description.

[Q] Quit: exits the Code mode.

[R] can be used to Rename code "n" from the map by entering a different description. Confirm with [Y].

[S]n_i gives output of a Set of "i" code numbers (max. 30) in the order specified by the user. End of set indicator: 999. The user can edit the sequence and select "concatenated" output (no [Y] confirmation between codes). Set contents are kept when leaving the code mode.

[T] performs Type conversion from Mk1 to Mk2 (and above) code. This conversion changes Hex 'E0' command indicator bytes into Hex '00' and makes the MSB of each command equal to '1'.

[V] controls output Volume for synthesizer output. Press [+] to increment and [-] to decrement. Any other key exits volume control mode.

[Shift]+[V] controls output volume for DAC output. Otherwise the same as [V].

[X]n m eXchanges code numbers "n" and "m" in the code map. The offset addresses are not modified.

15. EPROM mode

The EPROM mode is accessed from the main command level by means of the [E] command.

Upon first entry into the EPROM mode the user is asked to specify the type of EPROM he wishes to use (2716, 2732, 2732A, 2764 or "XXXX"). A different type of EPROM may be selected at any time.
The EPROM type controls the default addresses and the programmer hardware of the built-in programmer. When "XXXX" is selected commands involving the built-in EPROM programmer of the SAB cannot be used.

EPROM data can be sent to an external programmer by downloading a "CODES" file using the DOS "COPY" command. Such a file contains the "code" memory contents in Intellec Hex format, a code readable for most programmers.
The settings of the COM-ports can be changed using the "MODE" command from DOS.

When you want to examine or modify the hexadecimal codes the E(xamine) command can be used. Code addresses are displayed as decimal numbers while the actual memory contents are displayed as a hexadecimal number.

Some EPROM commands ask for 3 addresses (or require you to accept the default values with [ENTER]):

1. the code memory (RAM) starting address of the section to be programmed in the EPROM,
2. the EPROM address where programming has to start,
3. the EPROM address where programming should stop.

Note: The use of the EPROM programmer is not limited to speech codes. Any EPROM of the proper type can be read and any section of the "code" memory (starting from address 1) can be programmed anywhere within an EPROM (from address 2 to address 3).

The Read and Write commands can be destructive and therefore require a [Y] confirmation. Writing of an EPROM can be aborted with the [Esc] key.

When an EPROM error occurs the action may be continued for the rest of the address range by pressing [C]. Other characters stop command execution.

15.1 EPROM mode commands

[B] performs a Bitcheck on the EPROM to test whether the corresponding bytes from the "code" memory can be programmed.

[E]n allows you to Examine and change "code" memory contents starting from address "n" (decimal). The selected code byte is indicated by "<-" and displayed as a hexadecimal number. It can be altered by typing in the desired Hex value. Pressing [Enter] stores the data into "code" memory and steps to the next byte.
The cursor keys can be used to scroll through the code. Any other key will exit the Examine mode.

[M] displays the contents of the code Map. Each code number is followed by its offset address, the code length and bit rate (bits/s), and a 20-character description.

[N] performs a Newcheck on the EPROM to find any already programmed locations.

[Q] Quit: exits the EPROM programmer.

[R] Reads the EPROM and stores its contents in "code" memory.

[T] permits the Type of EPROM to be selected by entering a number (1-5) corresponding to the desired type. A list of permitted types and corresponding numbers will be displayed. Type 5 is used for "other types" and will disable the built-in programmer when selected.

[V] Verifies whether the EPROM contents match the data in the specified section of the "code" memory.

[W] Writes the contents of the specified code memory section into the EPROM. This operation requires 52 ms per byte. When the contents of an EPROM are extended by adding codes to its original map only the added bytes are programmed.
Write EPROM can be stopped using the [Esc] key.

16. FILE mode

The speech File mode is entered from the main command level using the [F] command. It is intended to store speech data on disc. It can be used to build your own speech library.

File mode operations involving disk access are carried out on the default (sub)directory specified in the "pathname". For these commands the target disc drive must be loaded, otherwise a DOS error will result (see section 7.1).
The default "pathname" can be changed using the [P] command which is also accessible in the Parameter mode (via the FUNCTION menu) and the Option mode.

The speech File mode uses a special directory file (FDIR) containing 79-character speech file descriptions:
- file type (C = codes, P = parameters, F = free)
- file number (1..999)
- file size (byte count or frame count)

- date of creation
- description of contents (max. 59 characters).

The FDIR file must be created using the M(ake directory) command.

All files are ASCII-files. Filenames consist of the filetype character followed by the 3-digit file number (e.g. P002, C014). Code files are formatted as INTELLEC HEX (= MDS86) code. This permits the use of an external EPROM programmer (see Ch. 14 and 15).

The root directory of a diskette has only 64 or 112 directory entries (single or dual-sided). This severely limits the possibilities of storing speech files. Therefore it is advised to use subdirectories that have no limitation on the number of files except the disc size.
A disc may have several subdirectories, so for every version or project a separate subdirectory can be used.

One diskette can contain parameters of approximately 75 or 150 seconds of speech (single/dual sided). When short words or word fragments are stored one diskette may contain as many as 500 parameter files!
An obvious drawback of such a large directory is the long time required to list its contents and to find the directory entry at every file access. Therefore it is advised not to make a speech directory size much larger than 200 files.

In the File mode some commands from the Parameter and Code mode are duplicated to facilitate direct code generation from parameter files stored on disk.
The procedure for code generation from disk files is as follows:
- use the R(ead) command to copy the parameters from disk into the parameter "input" memory;
- copy the "input" parameters into the "edit" memory using the L(oad) command;
- generate compressed code in the "code" memory from the "edit" memory contents by using the C(ode) command.

The R(ead) command displays the header of the file to be read before asking a confirmation with [Y].
The W(rite) command prompts the user with the first available free file number. A different file number may be used instead. An existing file may be overwritten in which case a [Y] confirmation is required.
The W(rite) command asks for a file description of max. 59 characters and is only executed after a [Y] confirmation.

The Audio output commands from the Parameter mode are also available for checking memory contents by ear.
The "code" memory contents can also be output through the synthesizer using the B(unch) or S(et) commands from the Code mode.

The filetype indicator (PARAM/CODES) shows which type of data is being stored or read and also the memory involved (PARAM: "edit" or "input" memory, CODES: "code" memory).
Please note that CODES operations involve the entire "code" memory; both map and compressed data are stored in or read from a disk file.

16.1 File mode commands

[A]	Audio output of "edit" parameters between the "edit" memory's BEGIN and END pointers.
[Shift]+[A]	Audio output of "input" parameters between the "input" memory's BEGIN and END pointers.
[B]n m	gives output of a Bunch of code messages from number "n" to number "m", each requiring confirmation with [Y]. Output sequencing stops at any other key. When "n" is greater than "m" sequencing wraps around to number 0 when end-of-map is reached.
[C]	generates compressed Code from the parameter "edit" memory between the "edit" memory's BEGIN and END pointers.
[D]n	lists the speech file Directory on the selected pathname starting from file number "n". Type, number, date of creation, size, and contents description (header) of each file are displayed.
[H]n	permits changing the Header (description of the file contents) of speech file "n". A confirmation with [Y] is required before the command is executed.
[K]n	Kill removes file "n" from the speech directory and from disc. K(ill) needs a [Y] confirmation.
[L]	Load copies the "input" memory contents into the "edit" memory. Confirm with [Y].
[M]n	Make creates a directory file (FDIR) for "n" file entries on the default "pathname". When a speech directory file already exists the system will issue a warning and ignore the command. Otherwise the command is executed after a [Y] confirmation.
[P]	sets the default Pathname for speech file access (format: DRIVE:\SUBDIR\). Confirm with [Y]. For more detailed information see DOS manual.
[Q]	Quit: exits the File mode.
[R]n	Reads speech file "n" from disc using the default "pathname". The selected file type must agree with the data type of the file to be read. After confirmation with [Y] the file contents are copied to the parameter "input" memory (file type PARAM) or the "code" memory (file type CODES).
[S]n_i	gives output of a Set of "i" code numbers (max.30) in the order specified by the user. End of set indicator: 999. The user can edit the set and select "concatenated" output (no [Y] confirmation between codes).

[T] toggles the selected file Type between "PARAM" and "CODES".

[V] selects Volume control for synthesizer output. Press [+] to increment and [-] to decrement. Any other key exits volume control mode.

[Shift]+[V] selects volume control for DAC output. Otherwise the same as [V].

[W]n Writes speech data to disc file number "n" on the default "pathname". Data originate from the parameter "edit" memory (file type PARAM) or from the "code" memory (file type CODES). Parameters are stored from the "edit" memory's BEGIN to END pointers. In a CODES file the entire contents of the "code" memory are stored or if "free format" data are detected the user must specify the number of bytes to be stored in the file.

[X] eXchanges the contents of the parameter "input" and "edit" memories.

17. OPTION mode

In the Option mode some startup defaults can be changed and stored on disc in the configuration file (CONFIG2.PCF).
Also the available memory size and the corresponding maximum number of frames are displayed.

17.1 Option mode commands

[B] Switches the system Beeper on or off.

[C] toggles the Computer type between IBM-PC/XT/AT and Philips P3100.

[D] changes the system Date, which is used in the File mode when storing speech files on diskette.

[F]n sets the sampling Frequency of the SAB to "n" Hz. The permitted range is 6000 to 20000 Hz. For the PCF8200 the sampling frequency is 10000 kHz.

[G] Gets the system configuration from the configuration file (CONFIG2.PCF) on the OM8210 disc. Make sure that the disc is loaded in the system startup drive.

[I]n changes the Ieee/Iec-address of the SAB to "n" (default is 19). The hardware address jumpers must be adjusted to the specified value immediately after executing this command (see section 17.3).

[K] Keep stores the system configuration on the OM8210 disc in the "CONFIG2.PCF" file. Make sure that the disc is in the system startup drive.

[L] enables input Level control. Press [+] to increment or [-] to decrement. Any other key will exit level control mode.

[M] selects the Mode of operation to be either I(interactive) or B(atch). In Batch mode the user input is read from a batch file and all confirmations are disabled (see section 17.2).

[P] sets the Pathname for speech files (format: DRIVE:\SUBDIR\). Confirm with [Y]. This command is also available in the File mode. For more detailed information see the DOS manual.

[Q] Quits the Option mode.

[S] toggles the Spectrum display switch. When this switch is "on" the FFT power spectrum of each frame is displayed during analysis.

[V] selects Volume control for synthesizer output. Press [+] to increment or [-] to decrement. Any other key exits volume control mode.

[Shift]+[V] selects volume control for DAC output. Otherwise the same as [V].

17.2 Batch mode

The Batch mode of operation was introduced to permit large amounts of speech material to be processed in as short a time as possible. Together with the introduction of SAMPLES files this permits running the program without a human operator (e.g. during non-office hours).

The program will read all keyboard input from a batch file specified by the user at startup time.
User confirmations are disabled in Batch mode. Otherwise all user input is the same as in Interactive mode.

The program must be started from DOS using the following sequence:

"SP2 <(batch file name)"[Enter]

The batch file containing the required user input must be an ASCII text file and can be created using an editor program such as EDLIN or PC-WRITE™.
An annotated example of such a file ("BATCH.EXP") is supplied on the OM8210 diskette. The contents of this file are listed in Appendix C.

17.3 Changing the SAB hardware address

For gaining access to the hardware address jumpers the front cover of the SAB must be removed.

Note: Before opening the cabinet first switch off the power, disconnect the power cord and remove the IEC cable.

When the front plate of the SAB has been removed, slide the IEC interface card forward. The address jumpers are located near the IEC connector (see fig. 8).

After adjusting the jumpers to the desired address slide back the IEC interface card and press home firmly. Next re-install the front cover. insert the power cord and the IEC cable. Now switch on the power to the SAB.

When the speech program is still loaded, make sure it is in the main command level. Then press [Q] to re-initialize the SAB hardware.

The jumper positions correspond to the binary representation of the SAB address. Figure 8 shows the jumper setting for the default address of 19 (see Chapter 5).

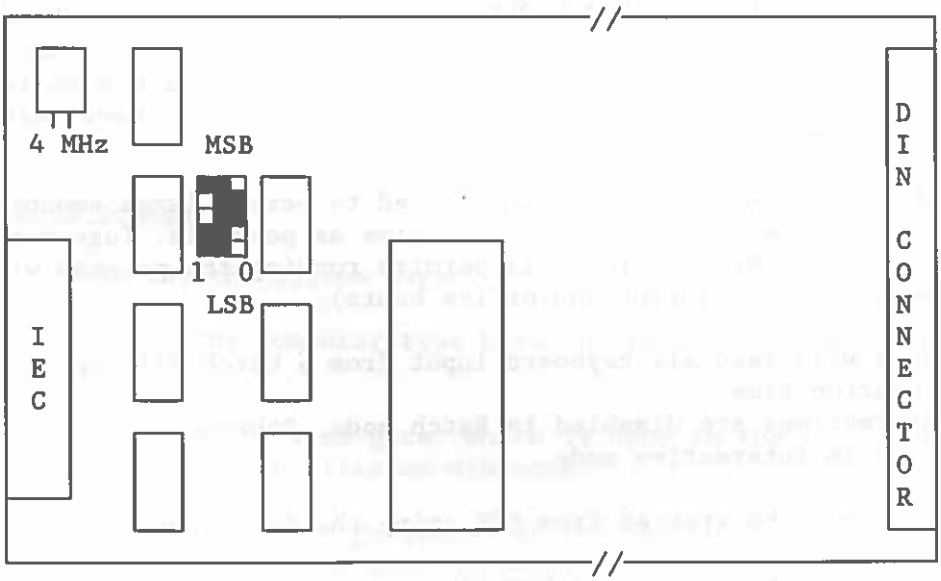

Figure 8. Address jumpers on the IEC control card

Appendix A: List of Pascal error codes

Code	Message
1	Hard Data Hard data error (parity, CRC, check sum, etc.).
2	Device Name Invalid unit/device/volume name, format, or number.
3	Operation Invalid operation: GET if EOF, RESET a printer, etc.
4	File System File system internal error, ERRS>15, etc.
5	Device Offline Unit/device/volume no longer available.
6	Lost File File itself no longer available.
7	File Name Invalid syntax, name too long, no temporary names, etc.
8	Device Full Disk full, directory full, all channels allocated.
9	Unknown Device Unit/device/volume not found.
10	File Not Found File itself not found.
11	Protected File Duplicate filename; write-protected.
12	File In Use File in use, concurrency lock, already open.
13	File Not Open File closed, I/O to unopen FCB.
14	Data Format Data format error, decode error, range error.
15	Line Too Long Buffer overflow, line too long.

Appendix B.1: Formant frequencies and bandwidths male speech

	values in Hz									
code	F1	F2	F3	F4	F5	B1	B2	B3	B4	B5
0	100	500	1500	2550	3900	3000	3000	3000	3000	3000
1	109	526	1690	2766	4600	600	800	600	700	800
2	119	554	1903	2999		303	433	190	265	335
3	130	583	2143	3253		153	234	60	100	140
4	141	614	2414	3528		77	126			
5	154	646	2719	3826		39	68			
6	168	680	3063	4149		20	37			
7	183	716	3450	4500		10	20			
8	199	754								
9	217	793								
10	237	835								
11	258	879								
12	282	925								
13	307	974								
14	335	1025								
15	365	1079								
16	398	1136								
17	433	1195								
18	472	1258								
19	515	1324								
20	561	1394								
21	612	1467								
22	667	1545								
23	727	1626								
24	793	1711								
25	864	1801								
26	942	1896								
27	1027	1996								
28	1119	2101								
29	1220	2211								
30	1330	2328								
31	1450	2450								

Appendix B.2: Formant frequencies and bandwidths female speech

	values in Hz									
Code	F1	F2	F3	F4	F5*	B1	B2	B3	B4	B5*
0	100	500	2050	3500	----	3000	3000	3000	3000	----
1	110	529	2251	3662	----	700	1000	700	800	----
2	121	561	2473	3831		414	626	265	322	----
3	134	594	2715	4007		245	391	100	130	----
4	147	629	2982	4192		145	245			
5	162	666	3275	4386		86	153			
6	179	705	3597	4588		51	96			
7	197	747	3950	4800		30	60			
8	217	791								
9	239	837								
10	263	886								
11	290	939								
12	319	994								
13	351	1053								
14	387	1115								
15	426	1180								
16	469	1250								
17	517	1323								
18	569	1401								
19	627	1484								
20	691	1571								
21	761	1664								
22	838	1762								
23	923	1866								
24	1017	1976								
25	1120	2092								
26	1234	2216								
27	1359	2346								
28	1497	2484								
29	1649	2631								
30	1816	2786								
31	2000	2950								

* this section is not used by female speech.

Appendix C: Listing of BATCH.EXP (example of batch file)

```
rem OM8210 example batch file   version: 1.2    date: 22.4.88   HWZ
rem
rem This file is for explanatory purposes only. It contains the elements
rem of a batch file which you may need for your own activity.
rem In order to use this file all "rem"-statements, trailing spaces and
rem empty lines must be removed before starting the activity.
rem The resulting file is referred to as BATCH.BAT below.
rem
rem To start a batch job from DOS use the following statement:
rem
rem             "SP2 <BATCH.BAT"[Enter]
rem
rem The standard input will now be taken from file BATCH.BAT rather than
rem from the keyboard.

OMBPA:\         rem Option: mode=batch (no confirm required),
                rem         set path for filer
QCI16           rem Code: create code map for 16 (or as many as needed)
entries


QSR1            rem Quit,Sample: read sample file nr. 1

QAMPFW FILE 1   rem Quit,Analysis: M(ale) tables, Parameter (automatic load)

                rem Funct-Writefile: save parameters
FA FW FILE 1A   rem Funct-Auto_FD, <space> to accept default
                rem FRAME_COST, Funct-Writefile: save parameters (AFD)

CFILE 1A        rem create PCF code of parameters (AFD)


QSR2            rem Quit,Sample: read sample file nr. 2

QAFPFW FILE 2   rem Quit,Analysis: F(emale) tables, Parameter (automatic
load)
                rem Funct-Writefile: save parameters
FA FW FILE 2A   rem Funct-Auto_FD, <space> to accept default
                rem FRAME_COST, Funct-Writefile: save parameters (AFD)

CFILE 2A        rem Code: create PCF code of parameters (AFD)


QF TW CODES-A   rem Quit,File: switch filetype to CODES, save map of PCF
codes
QD              rem Quit, exit to DOS
```

Appendix D: Instructions for 220 to 110 Volt conversion

First disconnect the SAB power cord and IEC cable. Then remove the front cover to gain access to the power supply unit.

Slide the power supply forward and remove it from the SAB cabinet. The 220 to 110 Volt conversion can be carried out without dismantling the power supply unit.
Place the unit on its side cover which is closest to the pc-board, the connector facing you.
This is the position of the board as depicted in fig. 9 below.
The unit can now be wired for 110 volt operation by soldering a connecting wire between tags x0101 and x0102 on the left edge of the pc-board.

After completing the conversion carefully slide the power supply back into the SAB cabinet. Applying too much force will cause the connector pins to bend or break.
Finally, re-install the front cover and connect the power cord and IEC cable.

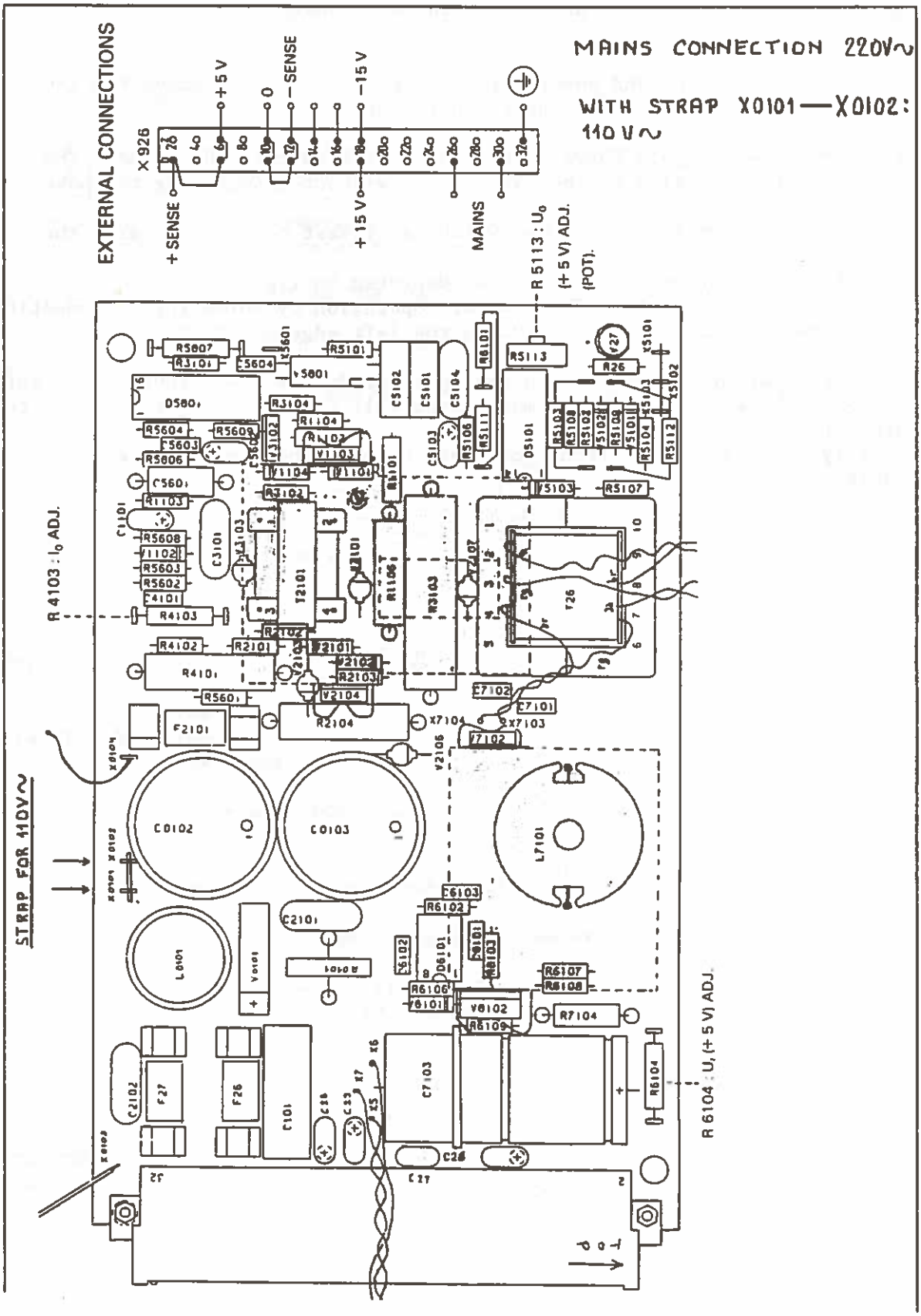

Fig. 9 Power supply connections

Components from the worldwide Philips Group of Companies

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Australia: PHILIPS INDUSTRIES LTD., Elcoma Division, 11 Waltham Street, ARTARMON, N.S.W. 2064, Tel. (02) 439 3322.
Austria: ÖSTERREICHISCHE PHILIPS INDUSTRIE G.m.b.H., UB Bauelemente, Triester Str. 64, 1101 WIEN, Tel. (0222) 60 101-820.
Belgium: N.V. PHILIPS PROF. SYSTEMS – Elcoma Div., 80 Rue Des Deux Gares, B-1070 BRUXELLES, Tel. (02) 52 56 111.
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Hong Kong: PHILIPS HONG KONG LTD., Elcoma Div., 15/F Philips Ind. Bldg., 24-28 Kung Yip St., KWAI CHUNG, Tel. (0)-24 51 21.
India: PEICO ELECTRONICS & ELECTRICALS LTD., Elcoma Dept., Band Box Building, 254-D Dr. Annie Besant Rd., BOMBAY – 400025, Tel. (022) 4930311/4930590.
Indonesia: P.T. PHILIPS-RALIN ELECTRONICS, Elcoma Div., Setiabudi II Building, 6th Fl., Jalan H.R. Rasuna Said (P.O. Box 223/KBY) Kuningan, JAKARTA 12910, Tel. (021) 51 7995.
Ireland: PHILIPS ELECTRICAL (IRELAND) LTD., Elcoma Division, Newstead, Clonskeagh, DUBLIN 14, Tel. (01) 693355.
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Netherlands: PHILIPS NEDERLAND, Marktgroep Elonco, Postbus 90050, 5600 PB EINDHOVEN, Tel. (040) 78 37 49.
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United Kingdom: PHILIPS COMPONENTS Ltd., Mullard House, Torrington Place, LONDON WC1E 7HD, Tel. (01) 580 6633.
United States: AMPEREX ELECTRONIC CORP., 230 Duffy Avenue, P.O. Box 560, HICKSVILLE, L.I.N.Y. 11802, Tel. (516) 931-6200.
(Active Devices & Materials) AMPEREX SALES CORP., Providence Pike, SLATERSVILLE, R.I. 02876, Tel. (401) 762-9000.
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