Realization of a real-time 16 kbit/s speech coder and decoder on a single digital signal processor chip each

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Award date:
1987

DEPARTMENT OF ELECTRICAL ENGINEERING
EINDHOVEN UNIVERSITY OF TECHNOLOGY
TELECOMMUNICATIONS DIVISION EC

REALIZATION OF A REAL-TIME 16 KBIT/S SPEECH CODER AND DECODER ON A SINGLE DIGITAL SIGNAL PROCESSOR CHIP EACH

by R.P.J. Kleuters

Report of the graduation work
accomplished from 8-9-1986 to 17-7-1987.
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SOURCE FILE LISTINGS:

SUMMARY

This report describes the design, implementation and testing of a real-time 16 kbit/s speech coder and decoder. The algorithm used is simple enough for the coder and decoder to be implemented each on a single digital signal processor chip (TMS32010).
In the transmitter the up to 4 kHz bandlimited speech signal is first split into eight mutually exclusive subband signals, each having a spectral bandwidth of 500 Hz. The sampling rates of the subband signals are reduced from 8 kHz to 1 kHz (being the Nyquist rate). For this subband splitting and sampling rate reduction, a very efficient technique called Quadrature Mirror Filtering is used. Next, each subband signal is independently coded according to its perceptual contribution to the overall subjective quality. For coding the subband signals, Pulse Code Modulation, with backward adaption of the quantization stepsize, is employed. Furthermore for each subband signal, the number of code bits is determined by a semi-adaptive bit-allocation algorithm. Finally, the coded subband samples together with the necessary side information and synchronization bits are multiplexed into a 16 kbit/s serial data stream.
In the receiver, the inverse operations (demultiplexing, decoding and speech signal reconstruction) take place.
The quality performance of the implemented Subband Coder, in particular the intelligibility of the reconstructed speech, is acceptable for our purposes. There is only little difference with unprocessed up to 3 kHz bandlimited speech. Furthermore, the capacities of each digital signal processor are nearly completely utilized by the implemented Subband Coder and Decoder algorithm, which means that an appropriate type of digital signal processor has been chosen.

LIST OF ABBREVIATIONS AND SYMBOLS

AIB	Analog Interface Board; part of Texas Instruments development system.
A/D	Analog to Digital (conversion).
AQB	Adaptive Quantization Backward.
CRT	Cathode Ray Tube.
D/A	Digital to Analog (conversion).
d_c	Stepsize adaption control factor.
d(n)	Log $\Delta(n)$.
$\Delta(n)$	Stepsize for (de-)quantization of a subband sample.
DPCM	Differential PCM.
DSP	Digital Signal Processor.
EVM	Evaluation Module; part of Texas Instruments development system.
F	Number of bits in a frame of the multiplexed signal.
FAW	Frame Alignment Word (in multiplexing operations).
FIR	Finite Impulse Response (filter).
f_s	Sampling rate.
γ	Stepsize adaption leakage factor.
I(n)	Two's complement B-bit code word; coded subband sample.
I/O	Input/Output.
m	Number of bits in a FAW.
M(.)	Stepsize adaption parameter.
m(.)	Log M(.).
p	Bit error rate.
PC	Personal Computer.
PCM	Pulse Code Modulation.
PDF	Probability Density Function.
PROM	Programmable Read Only Memory.
QMF	Quadrature Mirror Filter.
Qx	Fixed-point format in which a number is represented in the TMS32010. A number is represented with 1 sign bit, 15-x integer bits and x fractional bits.

RAM	Random Access Memory.
rms	Root mean square.
SBC	Subband Coder.
$s(n)$	8 kHz digital speech input to transmitter.
$\hat{s}(n)$	8 kHz reconstructed digital speech output from receiver.
$s(t)$	Analog speech input to transmitter.
$\hat{s}(t)$	Reconstructed analog speech output from receiver.
t_1	Mean time to acquire frame alignment.
t_2	Mean time between false indications of lost frame alignment.
$x(n)$	Quantizer input; uncoded subband sample.
$\hat{x}(n)$	Quantized subband sample.

1. INTRODUCTION

In the Telecommunications Division of the Eindhoven University of Technology work is done on a cooperative project with the University of Dar-es-salaam. This project concerns the distribution of audio-visual information via satellite channels to be used for tele-education, information dissemination and news distribution to the rural population and to isolated institutions in Africa. To reduce costs, services suitable for standard narrowband (64 kbit/s) distribution channels are necessary. Examples are transmission of still pictures plus speech, teletext plus speech, silent-teletext and scribophone. For more information on the development project, see [1].

This report deals with the realization of a part of the above mentioned sevices, namely a real-time speech coder and decoder. In this project 16 kbit/s of a 64 kbit/s channel is reserved for the transmission of speech. Another design requirement is to keep the costs as low as possible. Considering the enormous advances in digital technology, coupled with the increasing economics of digital hardware, it was decided to implement the coder and decoder in software on one digital signal processor (DSP) chip each. A technique that can allow this and that can realize a fairly good quality of the reconstructed speech at the receive side is known as Subband Coding [2,3,4,5].

As the first work on the Subband Coder (SBC) project a study has been made about the kind of Subband Coding which best meets the requirements [6]. From this work a proposal resulted for the type of Subband Coding to be used. During further work [7,8] implementations of functional blocks of this proposal were designed and tested.

To realize a properly working Subband Coder and Decoder these implementation designs had to be adjusted and extended. To meet the DSP capacities the software also had to be optimalized. Besides these alterations of the implementation designs and the software, the hardware communication between the transmitter and the receiver system had to be realized.

In this report the ultimate implementation design and hardware environment of a properly working real-time Subband Coder and Decoder realization are described.
First some general design considerations are discussed (chapter 2). This is followed by a description of the functional blocks of the Subband Coder and Decoder and the work that is done on the implementation design of them (chapter 3 to 6). After that the employed hardware will be considered (chapter 7). Finally the utilization of the transmitter and receiver DSP is discussed (chapter 8), followed by some suggestions for possible performance improvement (chapter 9) and the conclusions (chapter 10).

2. GENERAL DESIGN CONSIDERATIONS

2.1. Introduction

In this chapter some general aspects concerning the design of a Subband Coder are considered. First we will discuss the principle of Subband Coding. Then some features of the used digital signal processor are showed. This is followed by a description of the development equipment, and finally the design strategy is outlined.

2.2. General principle of Subband Coding

A general block diagram of a Subband Coder and Decoder is shown in Fig. 1.

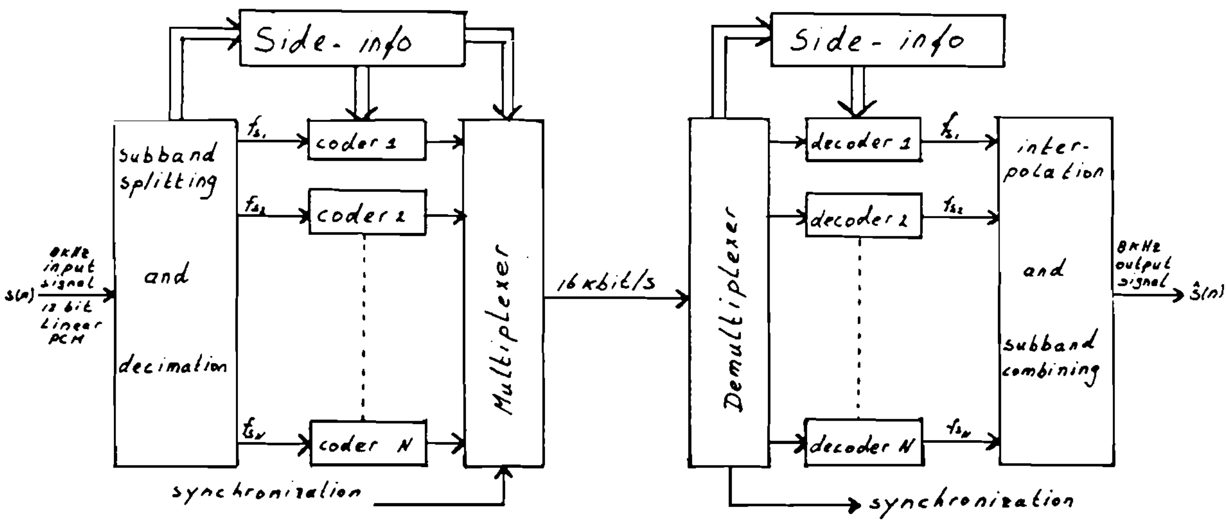


Fig. 1: General block diagram of a Subband Coder and Decoder.

The digital input to the transmitter, s(n), is obtained by sampling a bandlimited (up to 4 kHz) analog speech signal at a rate of 8 kHz, followed by 12 bit linear PCM analog-digital conversion. This signal is filtered into a number of N subband signals, each with a different spectrum that is part of the baseband frequency range 0-4 kHz. After the bandsplitting the sampling rates, f_s, of these subband signals can be reduced (decimation). This is possible because

the spectrum of each subband signal is narrower than the spectrum of the input signal. The minimum sampling frequency is twice the bandwidth of a subband signal. Decimation by an integer factor M can be achieved by retaining only one out of every M filter output samples. Then the subband signals are independently quantized and PCM coded into a number of bits that is determined by the bit-allocation used. In any case the bit-allocation is so as to allow an ultimate bit rate of 16 kbit/s. Finally the different subband signals together with the necessary side and synchronization information will be multiplexed into a serial 16 kbit/s data stream, representing the output signal of the transmitter.
In the receiver the inverse actions will take place. First the incoming data stream is demultiplexed to recover the coded subband signals plus the side and synchronization information. Next the subband signals are decoded in the PCM decoders. The decoding takes place according to the bit-allocation used, which in an adaptive case can be extracted from the received side information. By inserting zero-valued samples between the subband samples, followed by a filtering identical to that in the transmitter, the sampling rate in the subband signals will be restored to the original sampling rate of 8 kHz. During the filtering interpolation takes place, which gives the inserted zero's an appropriate value. Increasing the sampling rate with an integer factor M is accomplished by filling in M-1 zero-valued samples between each pair of filter input samples. By combining these final obtained subband signals the original digital speech signal, $\hat{s}(n)$, can be reconstructed.

The main advantages of Subband Coding relative to other coding techniques are:

- A low bit rate.
- The quantization noise generated within each subband remains confined to that channel and is independent of noise produced in other bands. In this way high level, low frequency quantization noise does not mask low level, high

frequency sounds, and vice versa.

- Each subband signal may be coded independently according to the perceptual contribution that each band makes to the overall subjective quality. This together with the sampling rate reduction in the subbands is the explanation for the low bit rate that can be achieved.
- By proper design a real-time implementation of the whole SBC principle is possible on two DSP's; one for the transmitter and one for the receiver.

These are enough reasons for using the Subband Coding technique for the realization of a 16 kbit/s speech coder and decoder that together deliver fairly good quality speech at a fairly low complexity (c.q. costs).

2.3. Capacities of the used DSP

The DSP used for the implementation of the SBC algorithm, is the TMS32010 from Texas Instruments. This TMS32010 is very suitable for speech processing applications. For an extensive documentation of the TMS32010, see [9]. As both the coder and decoder had to be implemented on one DSP chip each, it is not strange that the features and capacities of the TMS32010 have had a great influence on the design work. The most important ones will be discussed below.

2.3.1. Processor speed

The duration of one instruction cycle is 200 ns. Subband Coding is a real-time speech coding technique that in our case every 0.125 ms uses one input sample in the transmitter and creates one output sample in the receiver. So in the transmitter as well as in the receiver the whole coder resp. decoder program has to be passed through in 0.125 ms. So the longest cycle in either program may not count more than 625 instruction cycles.

2.3.2. Program memory and data memory

Program memory consists of up to 4K words of 16-bit width. In our case this concerns off-chip RAM. The available program memory is large enough and has been no constraint on the design work.
On the other hand data memory consists of 144 words of 16-bit width of RAM present on-chip. So data RAM is far too small to contain all variables for the coder or decoder. This problem can be solved by storing some data operands in off-chip program memory and then read them into on-chip data memory when needed. When they are not needed anymore they can be written back to program RAM if necessary. However this solution will slow down execution as reading from and writing to program memory each take 600 ns, whilst most instructions only need 200 ns. So it is necessary to carefully determine which variables are directly stored in data memory and which are stored in program memory. Variables that are not often used and tables for example can best be stored in program memory. For more information, see [7, pp. 39-40]. The available data memory has appeared to be a great constraint in the design of the SBC algorithm.

2.3.3. Arithmetical operations

Addition and subtraction are standard tools for the TMS32010 and will cost only one instruction cycle. Also multiplication can take place in 200 ns due to the presence of a 16*16-bit parallel multiplier.
However, the TMS32010 does not have an explicit divide instruction. A division therefore will have to be broken down into a series of subtracts and shifts with the consequence of a long execution time. So divisions have to be avoided as far as possible.

2.3.4. Fixed-point arithmetic

Computation on the TMS32010 is based on a fixed-point two's complement representation of numbers. Each 16-bit number is evaluated with a sign bit, i integer bits, and 15 minus i fractional bits. Thus the number :

```
0000 0010 1010 0000
         |
         |
         └──────decimal point
```

has a value of 2.625. This particular number is said to be represented in a Q8 format (8 fractional bits). Its range is between -128 (1000 0000 0000 0000) and +127.996 (0111 1111 1111 1111). The fractional accuracy of a Q8 number is about 0.004 (one part in 2**8 or 256).

To reduce quantization noise on the one side and to avoid overflows on the other side, it is very important to select an appropriate representation (i.e. Q-format) for each variable and constant.

On the design work we have started from the principle that the transmitter DSP input and the receiver DSP output have been normalized to the range between -1 and +1 Volt, and therefore a Q15 format has been taken for the representation of these two signals.

2.4. Equipment for testing and realizing the software developments

Two development systems as shown in Fig. 2 are available, one for the transmitter DSP and one for the receiver DSP. Such development system consists of two printed circuit boards, called "Evaluation Module" (EVM) and "Analog Interface Board" (AIB), and a host computer, in our case a Personal Computer from IBM.

The basic purpose of the Evaluation Module and the Analog Interface Board used is to enable the user to develop programs for the TMS32010 digital signal processor and to run

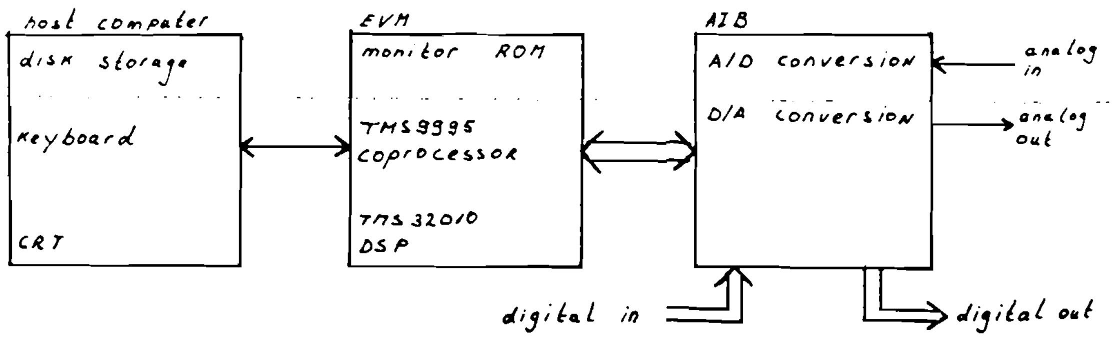


Fig. 2: Development system for the TMS32010.

these programs real-time. The programs can communicate with the analog "outside world" by means of A/D and D/A converters and bandlimiting/interpolation filters resident on the AIB. The program storage, the program design tools and the DSP itself are resident on the EVM board.

The two printed circuit boards of Texas Instruments are controlled by the host computer (IBM PC). This PC can communicate with the EVM via a serial interface. At the IBM PC the serial interface is controlled under BASIC. The EVM board is equipped with two serial interfaces of which only one, called "port 1", will be used. All serial interfaces are bi-directional. The EVM board together with the IBM PC form a program development system with which the user may enter assembler source files, assemble files, run them, execute them with breakpoints and single step programs.

The EVM and the AIB together can form a stand-alone system once these are programmed and started.

The EVM and AIB are well documented in [10] resp. [11]. In [7, pp. 24-32] the possibilities of the development system and the implications of this for a communications program resident in the IBM PC are discussed in more detail.

Later on in the project the development equipment has been extended with a so-called cross-assembler, which allows us to assemble source files on the IBM PC. This cross-assembler is described in [12] and [13].

2.5. Design strategy

When we look back at the general SBC principle described in 2.2., we can discern the following functional blocks:

- -filtering
- -bit-allocation
- -coding and decoding
- -multiplexing and demultiplexing

As a logical consequence the design work, implementation and testing has been done in a modular way.
First the filtering has been realized and tested on proper working. After that the bit-allocation has been added. This was followed by adding the coding and decoding. Finally everything has been supplemented with the multiplexing and demultiplexing.

3. ANALYSIS/SYNTHESIS FILTER BANK

3.1. Introduction

An important and critical aspect of the SBC design is the filter bank and its interaction with the sampling rate reduction (decimation) in the transmitter and the sampling rate increase (interpolation) in the receiver. A good reconstruction of the speech signal, $\hat{s}(n)$, requires a subband splitting in the transmitter and subsequent subband combining in the receiver with an overall frequency response that in magnitude equals 1 as best as possible.
If we leave out of consideration effects such as quantization noise, which are no direct consequence for filtering, there are two effects that can cause the magnitude of the overall frequency response to be not exactly equal to 1 everywhere. Both effects concern the fact that the filters we deal with in practice are not ideal in their frequency response, but contain beside the passband also a transition band and a stop band.

-The reduction of the subband sampling rates is necessary in order to maintain a low (16 kbit/s) bit rate in encoding these signals. This sampling rate reduction introduces aliasing terms [14, pp. 304-305] in each of the subband signals.

-In the reconstruction process the subband signals are combined together.As interpolation in the receiver takes place with a filtering identical to the one in the transmitter, after reconstruction the original signal has twice passed the filter bank represented by the frequency responses $H_1(f)$, $H_2(f)$, ...,$H_N(f)$. So exact reconstruction requires: $|H_1(f)|^2+|H_2(f)|^2+\ldots+|H_N(f)|^2\equiv 1$. In places were this requirement is not met, a ripple occurs in the response, called reconstruction ripple. Reconstruction ripple mainly occurs in the transition regions of the filter responses.

Thus concerning the filter bank development, aliasing and reconstruction ripple have to be suppressed or avoided as much as possible to accomplish a good speech signal reconstruction.
In the following the design, implementation and testing of the subband splitting and reconstruction method that has been chosen for our SBC realization is discussed.

3.2. Design considerations

Several methods exist for the splitting of the SBC input signal into a number of mutually exclusive subbands [6].
Some methods [15,16,17] are based on using a parallel bank of bandpass filters, as shown in Fig. 3.

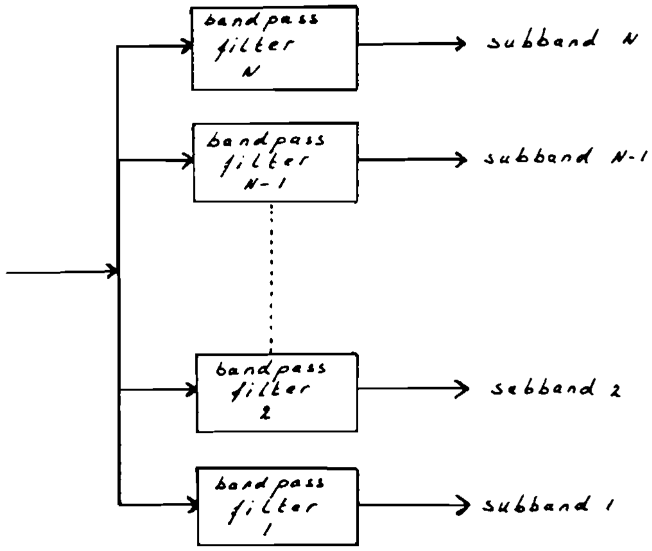


Fig. 3: Parallel bank of bandpass filters for subband splitting.

These filters may be conveniently realized by linear-phase Finite Impulse Response (FIR) networks. However, to meet the design requirements as mentioned in 3.1., very sharp transition band filters, i.e. very high order networks (ca. 200-tap) with a strong stop band attenuation (ca. 45 dB) are needed. Such filters would have a complexity that is far too high, excessive delay and possible performance limitations due to finite precision wordlengths if realized directly.

A more convenient alternative [18,19,20,21] is the use of a tree-structured filter bank. Such a tree configuration works by successively splitting the signal into two subbands at each branching point, using a high-pass/low-pass filter combination. The output from each intermediate filter is downsampled to the appropriate Nyquist rate for the signal and then applied to the next branching level for further spectral division, see Fig. 4.

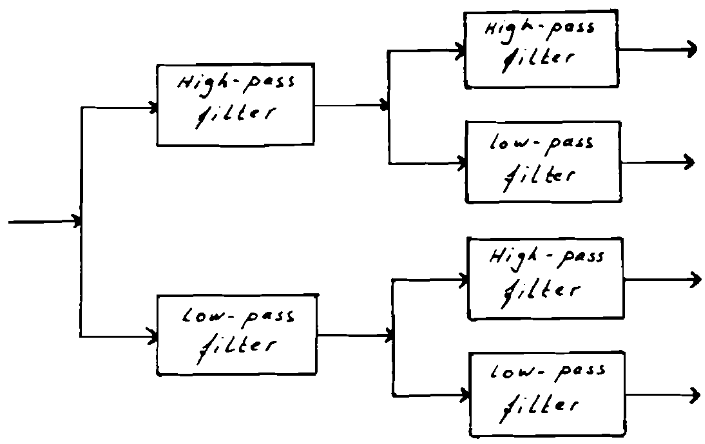


Fig. 4: Tree-structured filter bank for uniformly spaced subbands.

Besides this uniform subband splitting also octave-spaced subbands are possible when using the approach as shown in Fig. 5.

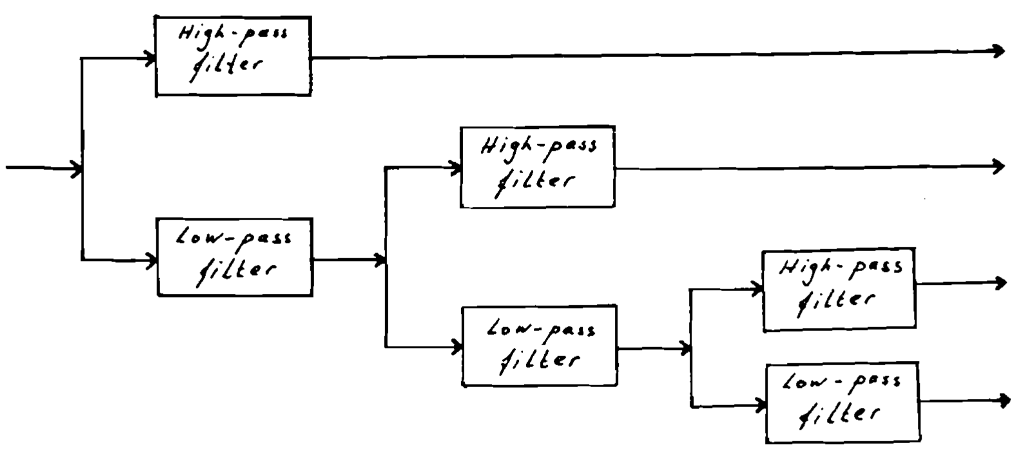


Fig. 5: Tree-structured filter bank for octave-spaced subbands.

If we would realize the tree-structured filter bank with the usual FIR filters, without taking any special measures, again very high order networks are needed.
However for the bandsplitting a technique called "Quadrature Mirror Filtering" can also be used. Quadrature Mirror Filters (QMF's) have special phase and magnitude characteristics which allow the splitting of a band into two equal width subbands and, upon reconstruction, provide for the cancellation of aliasing effects that occur during downsampling. Furthermore, as each band is splitted into two symmetrical parts, it is not very difficult to see to it that reconstruction ripple is strongly suppressed. Because of these properties, lower order filters (32 to 12 tap), which have fairly wide transition widths, can be used to cover the entire speech band of interest without any spectral gaps in the total response. Therefore, and also for some other reasons to be discussed later, a tree-structured QMF bank has been chosen for the realization of our SBC.

Fig. 6 reviews the basic configuration for a two-band SBC design that will be used for the explanation of the QMF bandsplitting and derivation of its design requirements.
The transmitter input signal s(n) is divided into two equally spaced frequency bands by low-pass and high-pass filters, $h_l(n)$ and $h_u(n)$, respectively. Each subband signal is reduced in sampling rate by a factor of two, i.e. if f_s is the sampling rate of the input signal, $f_s/2$ is the sampling rate of the subband signals. In practice the subband signals are then encoded and multiplexed for transmission, which in this (modular) design stage will be left out of consideration.
In the receiver the subband signals are interpolated back to their original sampling rates with the aid of similar low-pass and high-pass filters. The sum of the two interpolated subband signals, $\hat{s}(n)$, is the reconstructed version of the input signal, s(n).

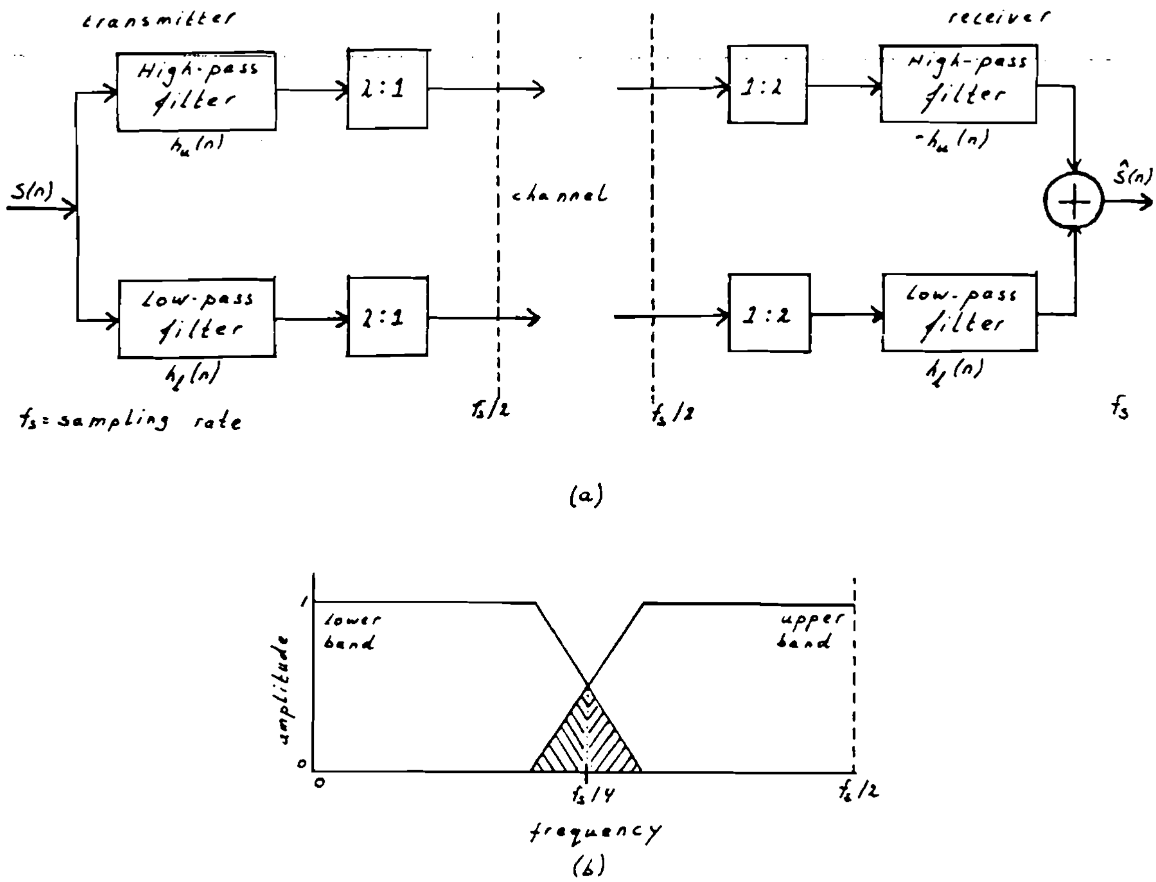


Fig. 6: (a) General block diagram of a two-band SBC (coding and multiplexing left out of consideration)
(b) Spectral description of the subbands.

To obtain the aliasing cancellation property, the filters $h_l(n)$ and $h_u(n)$ must be symmetrical FIR designs with even numbers of taps, i.e.,

$$h_l(n)=h_u(n)=0 \qquad \text{for } n<0 \text{ and } n>N-1 \qquad (1)$$

where N (even) is the number of taps. The symmetry property implies that

$$h_l(n)=h_l(N-1-n), \qquad n=0,1,2,\ \ldots,\ N/2-1, \text{ and} \qquad (2a)$$

$$h_u(n)=-h_u(N-1-n), \qquad n=0,1,2,\ \ldots,\ N/2-1 \qquad (2b)$$

Furthermore it is required that

$$h_u(n)=(-1)^n h_l(n) \qquad n=0,1,2, \ldots, N-1 \tag{3}$$

which is the mirror image relationship of the filters.
With the above constraints, the aliasing cancellation property of the QMF bank can be verified easily. A derivation is given in the appendix of [20].
To suppress reconstruction ripple as much as possible, the filters $h_l(n)$ and $h_u(n)$ ideally must satisfy also the condition

$$|H_l(f)|^2 + |H_u(f)|^2 \equiv 1 \tag{4}$$

where $H_l(f)$ and $H_u(f)$ are the Fourier transforms of $h_l(n)$ and $h_u(n)$ respectively. This also can be seen from the derivation in the appendix of [20].
The above filter requirement of eq. (4) cannot be met exactly except when N=2 and when N approaches infinity. However, it can be very closely approximated for modest values of N. Filter designs which satisfy eq. (2a) and approximate the condition of eq. (4) and the lowpass characteristic can be obtained with the aid of an optimization program. In practice "Hanning window" designs, optimized by the "Hooke and Jeeves algorithm" [6,21] will give satisfactory results. Fig. 7 shows the frequency response characteristics for an N=32-tap filter design, acquired with the above mentioned technique. As can be seen from Fig. 7b, the requirement of eq. (4) is satisfied to within ±0.025 dB, which is more than satisfactory for good SBC performance.

The QMF technique discussed for the two-band SBC from Fig. 6a can be applied in the same way at each branching point in a SBC that employs a tree-structured filtering into more than two subbands.

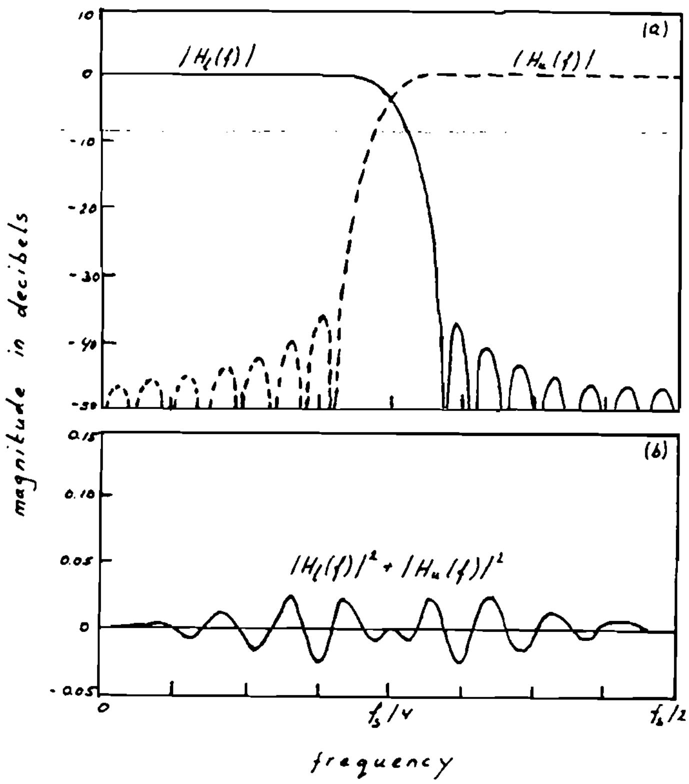


Fig. 7: Frequency response for a 32-tap QMF design [20]
(a) Magnitude responses of the individual filters
(b) Magnitude response of the composite system.

Finally we have to sketch the form of tree-structured QMF bank that has been chosen. As we will see in chapter 4, some adaption in the bit-allocation is necessary to achieve satisfactory reconstruction speech quality. However, if the subbands are unequally spaced, this requires techniques of a very high complexity. Therefore we have chosen uniformly spaced subbands [6,18]. Furthermore, the perceptual quality of the recovered speech will increase as the number of subbands increases. The number of subbands to be used in practice is a trade-off between recovered speech quality, processing complexity and delay in the filter bank. For our purposes an eight-band SBC has been found to represent a good compromise [6].
A sketch of the ultimate filterbank design to be implemented in the transmitter and receiver is shown in Fig. 8.

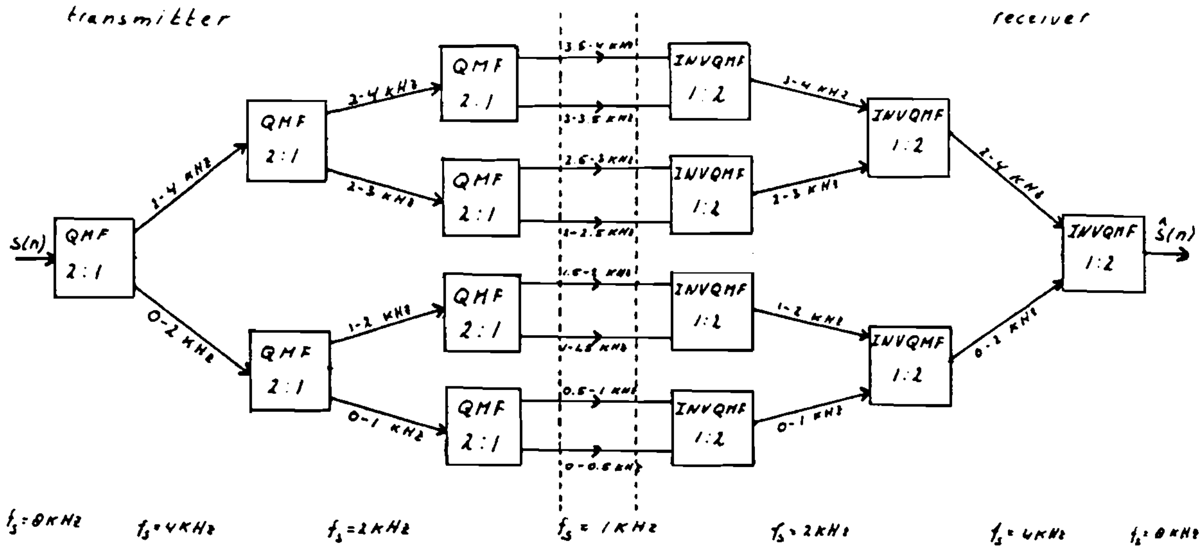


Fig. 8: Eight-Band subband splitting an reconstruction using a tree-structured QMF bank.

3.3. Implementation of the chosen filter bank algorithm

Here again, the basic principle of the implementation of the eight-band tree-structured QMF bank is discussed for the case of a two-band SBC, see Fig. 6a. From the mirror image property of the QMF bank described by eq. (3), we note that the coefficients used for the upper and lower subband filters are identical, except for the signs of the odd numbered coefficients. This property can be used to save a factor of two in computation by sharing the computation between the

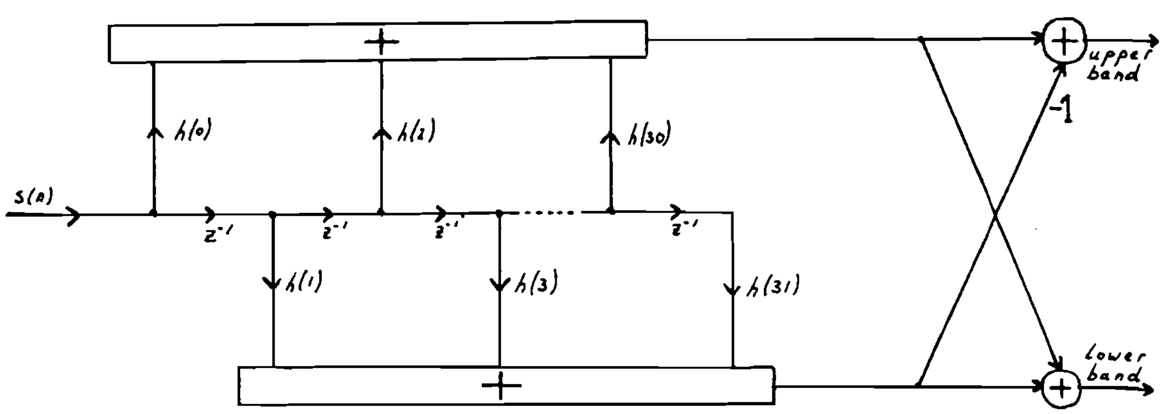


Fig. 9: QMF bank structure that shares computation between upper and lower filters.

filters in the manner described in Fig. 9, where $h_l(n)=h(n)$ and $h_u(n)=(-1)^n h(n)$. The partial sums of products are accumulated separately for the even- and odd-filter coefficient values. The sum of these two partial sums then gives the lower subband signal, and their difference produces the upper subband signal. Since the subband sampling rates are one-half of the input sampling rate, an additional factor of two is gained by computing the sums of products indicated in Fig. 9 once for every other input sample. Thus, each sample is shifted two delays in the shift register of Fig. 9 before being used.

Because of this sampling rate reduction, the filter structure of Fig. 9 can be divided into two parts as shown in Fig. 10.

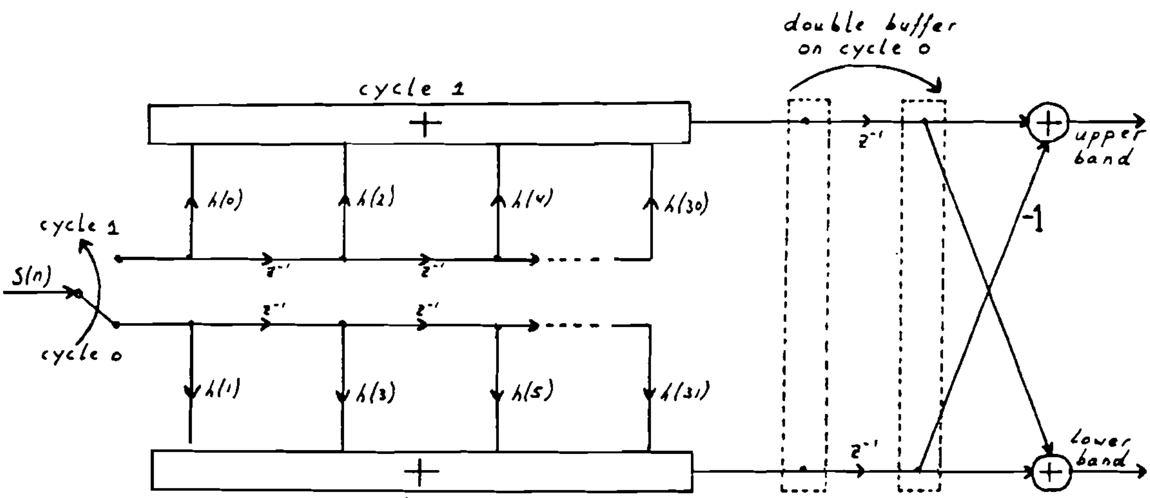


Fig. 10: Polyphase QMF bank structure of a SBC transmitter.

This structure is a two-band version of a more general class of multirate structures sometimes referred to as polyphase structures. As Fig. 10 shows, the input signal is separated into two sets by a commutator. Assuming that the commutator is in the lower position at time n=-1, the lower branch receives odd values of s(n), i.e s(-1), s(1), s(3), ..., and the upper branch receives even values of s(n), i.e. s(0), s(2), s(4), ... Both branches now operate at one-half of

the original sampling rate. Odd values of s(n) are filtered at odd sample times in the lower branch with an N/2 tap filter of odd valued filter coefficients (cycle 0). Similarly, even valued samples of s(n) are filtered in the upper branch with a N/2 tap filter of even filter coefficients. Furthermore double buffering is required when data computed in one cycle are needed in another cycle. The outputs of the even and odd filters are computed and stored in the left buffer for cycles 0 and 1. For taking the sum and difference the available data from the right buffer, which have been computed in the previous 0 and 1 cycles, are used. At the beginning of cycle 0 the data are transferred from the left buffer to the right buffer. The sum in the lower branch of Fig. 10 is computed in cycle 0, while the difference is computed in cycle 1.
A similar efficient polyphase structure can be generated for the QMF synthesis bank in the receiver. The resulting structure is shown in Fig. 11, that speaks for itself.

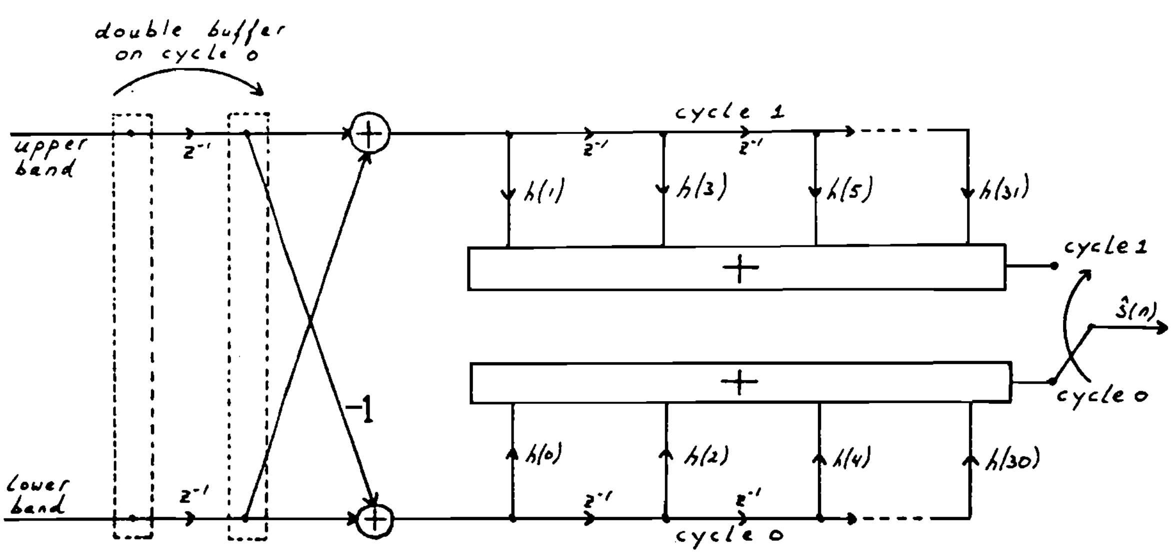


Fig. 11: Polyphase QMF bank structure of a SBC receiver.

So, in this QMF analysis and synthesis implementation, that makes efficient use of the available DSP resources, the computation load is evenly distributed between even and odd time cycles of the input sampling rate. This also has been

one of the reasons for choosing the QMF technique.
Table 1 reviews the two-cycle control structure that is used in the implementation of a two-band QMF splitting and reconstruction.

Table 1: Control structure for two-band QMF analysis and synthesis.

Cycle 0	A. Transmitter 1. Double buffer 2. Create lower band output by summation 3. Input one sample of s(n) 4. FIR filter (lower branch) B. Receiver 1. Double buffer 2. Create lower branch filter input by subtraction 3. Input one subband sample and store in left buffer 4. FIR filter (lower branch) 5. Output one sample of $\hat{s}(n)$
Cycle 1	A. Transmitter 1. Create upper band output by subtraction 2. Input one sample of s(n) 3. FIR filter (upper branch) B. Receiver 1. Create upper branch filter input by summation 2. Input one subband sample and store in left buffer 3. FIR filter (upper branch) 4. Output one sample of $\hat{s}(n)$

For the implementation of the eight-band SBC, this technique is repeated at each branching point of the filter tree. As the eight-band SBC is a multirate system with an ultimate sampling rate ratio of eight, it requires an eight-cycle control structure. For each cycle only one out of the eight possible paths in the QMF tree has to be passed through. A modulo-8 counter, called PATH, is used to assign the path in the QMF tree to be processed.

A sketch of the implemented QMF filter bank for the eight-band SBC (transmitter), including the subband processing sequence according to PATH, is shown in Fig. 12. For example, during cycle 3 (PATH=3) a sample of subband 5 (2000-2500 Hz) is created by respectively passing through HIGH1 in the first QMF stage, HIGH3 in the second stage and LOW7 in

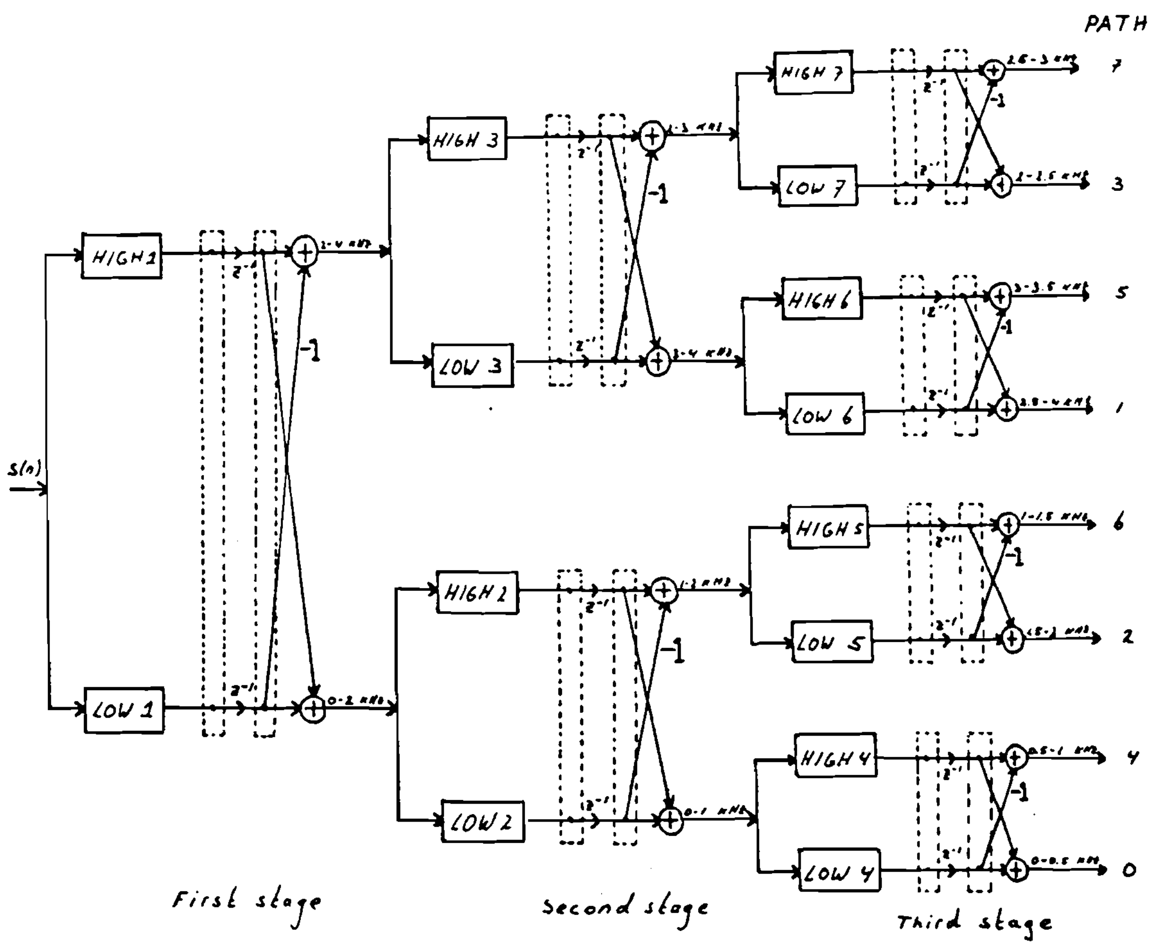


Fig. 12: QMF filter bank implementation for the eight-band SBC (transmitter)

the third stage. Also it has to be taken into account that after high-pass filtering the input signal into the 2-4 kHz part and sampling rate reduction, the 2-3 kHz part is obtained by high-pass filtering in stead of low-pass filtering. This can be verified easily by an examination of the spectra. Similar circumstances occur at some other parts in the QMF tree. As can be seen from Fig. 12, for PATH=0 to 7 the subband processing sequence is : 0-500 Hz, 3500-4000 Hz, 1500-2000 Hz, 2000-2500 Hz, 500-1000 Hz, 3000-3500 Hz, 1000-1500 Hz, 2500-3000 Hz, being respectively subband 1,8,4,5,2, 7,3,6.

For the first QMF stage 32-tap FIR filtering is used (16 taps for the upper branch and 16 taps for the lower branch). For the second and third stage 16- resp. 12-tap FIR filtering is used. The coefficients, as obtained by the Hooke and Jeeves optimization method [21] are depicted in Table 2.

Table 2: Filter coefficients for 12- ,16- and 32-tap QMF filters.

	12-tap
h(0) =h(11)=-0.3809699E-2	h(3) =h(8) =-0.8469594E-1
h(1) =h(10)= 0.1885659E-1	h(4) =h(7) = 0.8846992E-1
h(2) =h(9) =-0.2710326E-2	h(5) =h(6) = 0.4843894E+0
	16-tap
h(0) =h(15)= 0.1050167E-2	h(4) =h(11)=-0.9666376E-2
h(1) =h(14)=-0.5054526E-2	h(5) =h(10)=-0.9039223E-1
h(2) =h(13)=-0.2589756E-2	h(6) =h(9) = 0.9779817E-1
h(3) =h(12)= 0.2764140E-1	h(7) =h(8) = 0.4810284E+0
	32-tap
h(0) =h(31)= 0.6910579E-3	h(8) =h(23)=-0.4187483E-2
h(1) =h(30)=-0.1403793E-2	h(9) =h(22)=-0.3123862E-1
h(2) =h(29)=-0.1268303E-2	h(10)=h(21)= 0.1456844E-1
h(3) =h(28)= 0.4234195E-2	h(11)=h(20)= 0.5294745E-1
h(4) =h(27)= 0.1414246E-2	h(12)=h(19)=-0.3934878E-1
h(5) =h(26)=-0.9458318E-2	h(13)=h(18)=-0.9980243E-1
h(6) =h(25)=-0.1303859E-3	h(14)=h(17)= 0.1285579E+0
h(7) =h(24)= 0.1798145E-1	h(15)=h(16)= 0.4664053E+0

These coefficients are best represented in Q16 format (see 2.3.4.). In the beginning all time varying signals in the QMF filter bank were decided to be represented in the same format as the input signal s(n), i.e. in Q15 format. Later, as will be discussed in 3.5., these representations have been adapted according to the properties of speech spectra.
For reasons mentioned in 2.3.2., all filter coefficients (only 30, thanks to the symmetry property, eq. (2), of the QMF's) are permanently available in data memory. The same holds for the delay variables of the first QMF stage and furthermore for all buffer variables. However, the delay variables of the second and third QMF stage are stored in program memory. See also [7, pp. 39-40].
Appendix C contains a listing of the source file of the transmitter QMF bank, appendix H that of the receiver QMF bank.

3.4. Evaluation of the filter bank performance

Proper working of the analysis and synthesis filter bank implementation has been tested with the configuration as shown in Fig. 13.
The analog input, s(t), is bandlimited and subsequently converted to s(n) by 8 kHz sampling followed by A/D conversion. Next the transmitter DSP, loaded with the QMF analysis

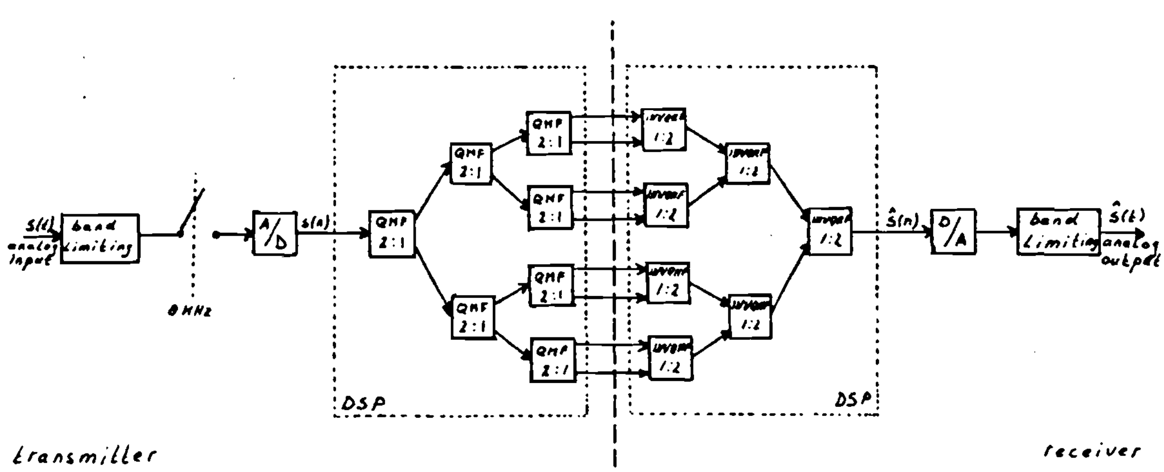


Fig. 13: Test configuration for the analysis and synthesis filter bank.

program, creates for each 8 kHz s(n) sample a 1 kHz subband sample. Each subband sample is transported in a parallel way from the transmitter DSP to the receiver DSP. In the receiver DSP, loaded with the QMF synthesis program, for each incoming subband sample a 8 kHz output sample, ŝ(n), is produced. By D/A conversion followed by bandlimiting, ŝ(n) is converted to the analog output ŝ(t).

How the bandlimiting and the A/D and D/A conversion are performed, will be discussed in chapter 7. The hardware realization and synchronization of the parallel communication between transmitter and receiver system will not be discussed, as it is not functional for the ultimate SBC realization.

The transmitter DSP program is controlled by interrupts. After each 8 kHz interrupt, created by the A/D converter, an interrupt service routine is executed. In this interrupt service routine a sample s(n) is read in and a created subband sample is written out. When the interrupt service routine has finished, the QMF program (for one out of eight paths) is carried out followed by a wait cycle for the next interrupt. QMF program plus wait cycle together form the main routine.

The receiver DSP program is also controlled by 8 kHz interrupts from the regained system clock, which in our circumstances is retained from the transmitter system (see also chapter 7). In the interrupt service routine a subband sample is read in and a reconstructed speech sample, ŝ(n), is written out. In the subsequent main routine, the inverse QMF program (for one out of eight paths) is carried out followed by a wait cycle for the next interrupt.

So far the same QMF principle also had been realized during previous work [7,8]. Therefore objective measurements of the filter bank performance give the same satisfactory results (for the reconstruction a "flat" curve in the band of interest) as are extensively discussed and illustrated in [7] and [8].

A subjective test has been carried out by applying an analog speech signal from a tape recorder to the transmitter input, and listening to the reconstructed speech signal via the audio output of the receiver system. By short-circuit of s(n) and ŝ(n), processed speech could be easily compared with unprocessed speech. The intelligibility of processed and unprocessed speech was quite the same. However, processed speech appeared to suffer more from quantization noise, due to the 16-bit representation used in the DSP's.

3.5. Improvement of the subjectively perceived quality

Till now, each time varying signal in the QMF tree has been represented in Q15 format. Watching a long-term speech spectrum as shown in Fig. 14, it is not difficult to see that in general the amplitude range will not be the same for each time varying signal in the QMF tree. So, as mentioned in 2.3.4., the quantization noise can be reduced by choosing an appropriate representation format for each signal in the QMF tree.

Extensive testing, being a very time consuming affair, resulted in the signal representation as depicted in Fig. 15. The signals in the receiver have to be represented in cor-

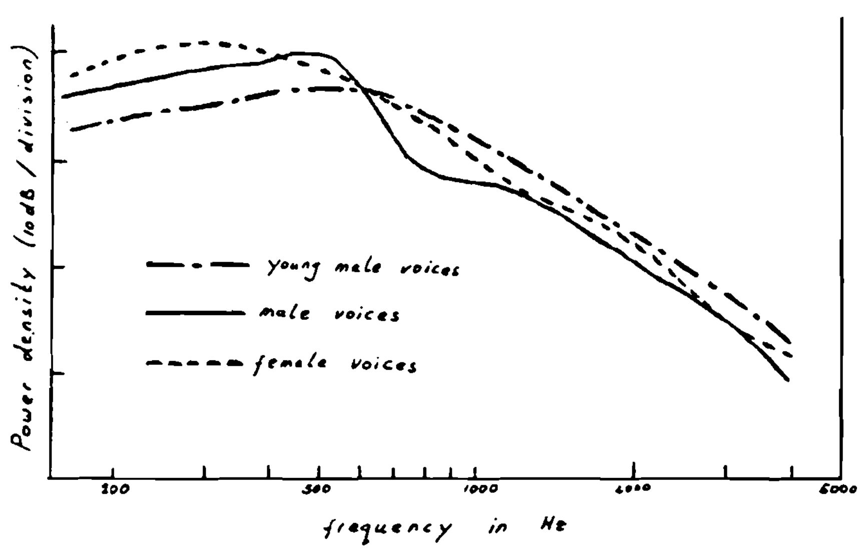


Fig. 14: Long-term spectrum of speech based on measurements by Beranek, Dunn and White [15].

responding formats. This representation has been implemented, performing a reconstructed speech quality that differs not or only marginally from unprocessed speech.
This adjustment of the subband signal representation will also turn out to be advantageous for the encoding of the subband signals (chapter 5), as the dynamic ranges to be covered are better specified.

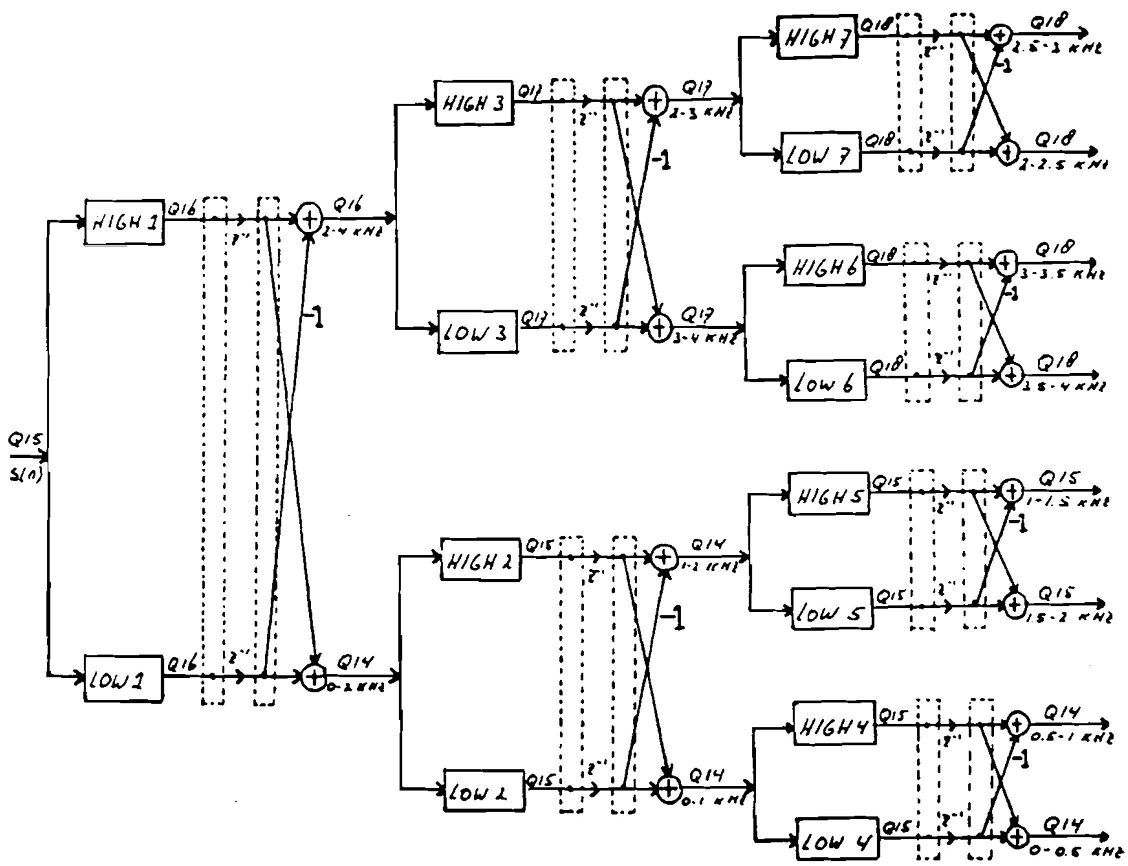


Fig. 15: Signal representation in (transmitter) QMF filter bank.

3.6. Conclusion

The QMF technique has proved to be very suitable for the implementation of the eight-band filter bank. We do not exaggerate unduly if we call $\hat{s}(n)$ a delayed replica of $s(n)$. However, the encoding of the subband signals into a 16 kbit/s bit stream, left out of consideration till now, will cause a performance degradation by adding an amount of quantization noise to the replica of $s(n)$.

4. BIT-ALLOCATION

4.1. Introduction

As discussed in 2.2., after the subband splitting each subband signal is coded independently into a number of bits. The ultimate bit rate has to be 16 kbit/s. So, if we reserve 1 kbit/s for side and synchronization information (see also chapter 6), 15 bits per ms are left to divide over the eight 1 kHz subband signals. This division of 15 bits over the eight subbands is determined by the bit-allocation. For realizing a proper quality it is desirable [6] that the number of bits, allocated to a subband, agrees with its perceptual contribution to the overall subjective quality. As can be seen from Fig. 14, this contribution will not be the same for each speaker. The same holds for one speaker at various moments. In other words, the bit-allocation, which is optimal at one certain moment and for one certain speaker, is not necessary the best at another moment and/or for another speaker.
In this chapter the design and implementation of the bit-allocation algorithm that has been chosen for our purpose is discussed. Also described is an experiment to test the functioning of the implemented algorithm.

4.2. Design considerations

Three methods exist [6] for allocating the bits to the several subbands.
For the first one, called fixed-bit-allocation, the number of bits allocated to a subband is the same at each moment for each speaker. This method will not perform an acceptable quality [3,23].
The second method, called dynamic bit-allocation, is a fully adaptive bit-allocation algorithm, where the power is measured in each band and the bits are successively allocated to the subbands [23]. Proper implementation of this method (in combination with the right coding algorithm) will lead to

very acceptable results. However, this method requires a lot of side information and delay and furthermore is far too complex to fit the DSP device capabilities.
Therefore a compromise solution, called semi-adaptive bit-allocation, has been chosen for the realization of the SBC. A semi-adaptive bit-allocation scheme is, compared to a fully adaptive bit-allocation method, relatively simple to implement, yet gives a significant improvement in performance over a fixed-bit-allocation scheme. Furthermore the required side information is reduced greatly and, as we shall soon see, the semi-adaptive algorithm does not introduce any additional delay over and above that of the QMF transmitter filter bank.

The first QMF filter outputs are used to obtain energy estimates for the bit-allocation algorithm. The spectral envelope estimate, approximated by the short-term-average magnitude, is computed during a window, determined by the average period for wich speech signals are stationary. This is done for the speech in each of the subbands, 0-2 kHz (L) and 2-4 kHz (H). At time intervals, dictated by framing considerations (chapter 6), the ratio between average magnitude estimates for the two bands is used to form a three way decision as to whether the speech is voiced, unvoiced or intermediate. The following voicing decision scheme, recommended in the literature [2,3,4], is used:

$L/H > 20$	voiced
$1.5 < L/H < 20$	intermediate
$L/H < 1.5$	unvoiced

Hysteresis is included in the decision-making process to prevent rapid strategy changes due to marginal decisions. If the voiced strategy is already in use, L/H must be less than 10 to change back to the intermediate strategy, and for the unvoiced strategy in use, L/H must be greater than 3 to revert back to the intermediate strategy.
The frequency range 200-3200 Hz is seen as the band of

interest [6,20] in a speech signal. The 0-200 Hz part contains a lot of power that however is not important for the intelligibility of speech. This part can even better be omitted by the bandlimiting filter (Fig. 13) before the QMF splitting takes place, since it has a wrong influence on the voicing decision and causes a lot of aliasing, both resulting in an inefficient coding of the subband signals. Also the part above 3200 Hz, containing only very little power (Fig. 14), is unimportant for speech intelligibility and may be omitted by the bandlimiting filter (as this part contains very little energy, the filtering required at this edge is not as sharp as at the 200 Hz edge). To cover the band of interest as best as possible, the bit-allocation assignment for the three possible voicing strategies as shown in Table 3 is used [2,3,4].

Table 3: Bit-allocation assignment.

band (kHz) to	0.0 0.5	0.5 1.0	1.0 1.5	1.5 2.0	2.0 2.5	2.5 3.0	3.0 3.5	3.5 4.0
voiced	4	4	3	2	2	0	0	0
intermediate	4	3	2	2	2	2	0	0
unvoiced	2	3	3	3	2	2	0	0

As can be seen from this table, no bits are allocated to the 3-3.5 kHz subband (i.e. 3000-3200 Hz). Using these bits to improve the coding of the other subband signals, results in a better reconstructed speech quality than inclusion of coding of the 3000-3200 Hz part will do. By consequence it is not necessary to split (and in the receiver to reconstruct) the 3-4 kHz band in the QMF filter bank. However, the 3000-3200 Hz part may not be omitted by the bandlimiting filter, since it contributes to the voicing characteristic of the speech signals.

Ultimately the voicing strategy used also has to be transmitted to the receiver. As there are only three possibilities, the side information required is modest.

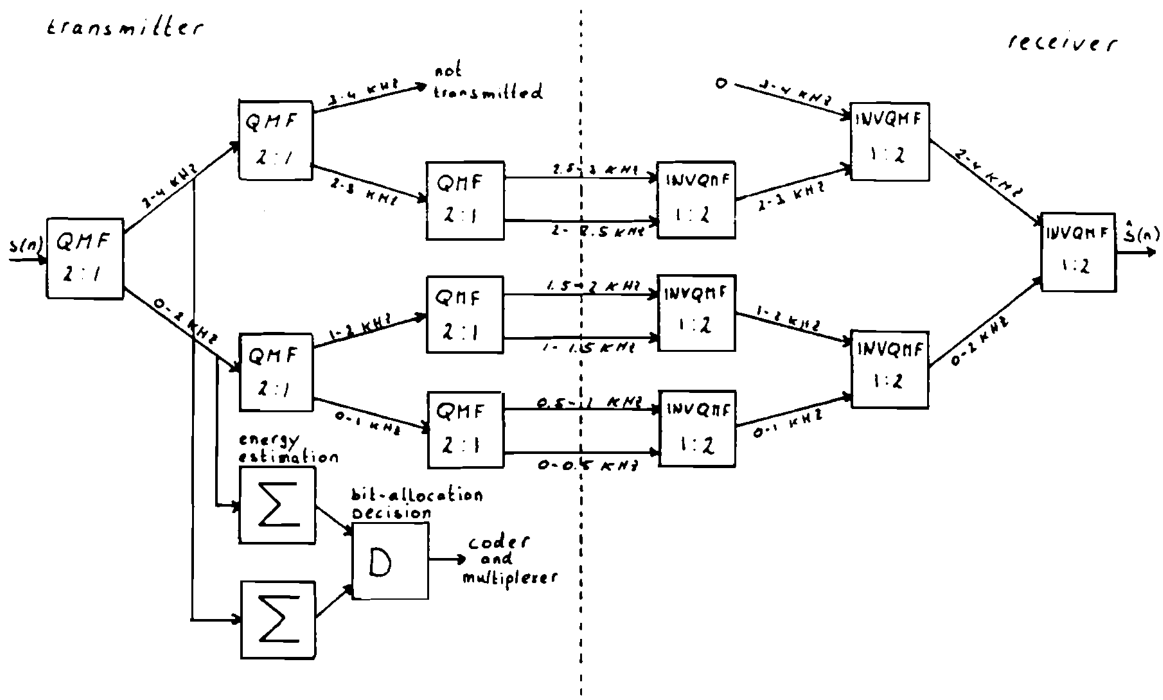


Fig. 16: Semi-adaptive bit-allocation in QMF filter bank.

The filter bank design, extended with the semi-adaptive bit-allocation to be implemented, is sketched in Fig. 16.

4.3. Implementation of the chosen bit-allocation algorithm

The energy summation and the voicing decision itself are straight implementations of the design discussed in 4.2. (using appropriate representation formats for the variables, all being directly stored in data RAM), and therefore need no further explanation. What really matters in the implementation of the bit-allocation algorithm is the synchronization with the rest of the SBC program.

For reasons to be discussed in chapter 6, a frame length of 128 bits (8 ms) is used in multiplexing the coded subband signals. Speech may be considered to be in a stationary state during a period of 8 ms. Therefore, each time a new frame starts, the voicing strategy is updated in agreement with an 8 ms energy measurement. As this energy measurement takes place in the first QMF stage, we have to take into account the processing delay time in the second and third stage of the QMF filter bank. This delay time is calculated to be 11.5 ms. As the energy measurement takes 8 ms, after the voicing decision it will last another 3.5 ms before the first samples corresponding with the voicing decision enter the PCM coders. Therefore the voicing decision takes place 3.5 ms before a new frame starts, at which the new strategy is taken over (buffering is required). As the 3-4 kHz subband is not transmitted and therefore not further splitted in the QMF filter bank, a lot of processing time is saved by executing such voicing decision, where otherwise the 3-4 kHz bandsplitting would occur.

To perform this synchronization a down-counter, called FRAME, is used in addition to the modulo-8 counter PATH, introduced in 3.3. PATH determines the subband to be processed for an 8 kHz cycle. As discussed earlier, for PATH=0 to 7 this is the subband sequence 1,8,4,5,2,7,3,6. For one frame this sequence has to be passed through eight times, kept up by FRAME. PATH is updated for each 8 kHz input, FRAME is updated each time just before a subband 1 sample is coded. At the beginning of a frame, the down-counter FRAME is loaded with the value 7. Each time when PATH=0, FRAME is adjusted. When FRAME=0 and PATH=0 the voicing strategy for the PCM coding is updated and a new frame is started. When FRAME=3 and PATH=5, being ca. 3.5 ms before a new frame start, a new voicing decision is made.

In the receiver the same synchronization principle is used. Here, no voicing decision has to take place. The voicing strategy for the PCM decoding is retained from the side information (available in the frame alignment word, chapter 6). As in the receiver the PCM decoding is done before the QMF reconstruction, FRAME is updated each time that PATH=7. A new frame is started and the voicing strategy is updated when FRAME=0 and PATH=7.

The whole synchronization is depicted in Fig. 17.

```
FRAME          7       6       5       4
PATH           01234567012345670123456701234567
SUBBAND        18452736184527361845273618452736
               update voicing strategy

FRAME          3       2       1       0
PATH           01234567012345670123456701234567
SUBBAND        18452736184527361845273618452736
                  voicing decision
```

(a)

```
FRAME          7       6       5       4
PATH           01234567012345670123456701234567
SUBBAND        18452736184527361845273618452736

FRAME          3       2       1       0
PATH           01234567012345670123456701234567
SUBBAND        18452736184527361845273618452736
                                               update voicing
                                               strategy
```

(b)

Fig. 17: (a) Frame synchronization in transmitter
(b) Frame synchronization in receiver.

The semi-adaptive bit-allocation software has been added to the filter bank programs of appendix C and H.

4.4. Experimental check

After the DSP programs had been extended with the semi-adaptive bit-allocation algorithms, some testing has been done with the same configuration as shown in Fig. 13. Again an analog speech signal was applied to the transmitter input and listening to the reconstructed speech signal took place via the audio output of the receiver. Although the PCM coding and decoding had not yet been implemented, the perceived quality of the reconstructed speech was not as good as in the case discussed in 3.5. This is a consequence of not transmitting the 3-4 kHz band. However, the quality, for our purpose particularly the intelligibility, was still fairly satisfactory. Therefore, during all further work this quality has been taken to be the maximum quality that can be achieved with our 16 kbit/s speech coder.

This time, also an analog signal, corresponding with the voicing strategy used, was retained from the transmitter system (AIB) and applied to an oscilloscope. In this way the changing of the voicing strategy for different "types" of speech has been confirmed. And indeed changes do not occur within a 8 ms period. Mostly the voicing strategy is constant for more than one 8 ms period, so the choice of 8 ms as a period during which speech signals may be considered stationary is adequate.

Furthermore, by setting breakpoints and single stepping (2.4.) also the synchronization has been checked.

The suitability of the semi-adaptive bit-allocation and the used bit assignments, shown in Table 3, can (apart from 0 bits allocated to the 3-4 kHz band) only be tested if the PCM coding and decoding has been implemented, and therefore will be discussed in 5.4.2.

5. CODING AND DECODING OF THE SUBBAND SAMPLES

5.1. Introduction

Before coding the different subband signals, an incoming sample or difference between an incoming sample and an estimation value of it (in case of differential PCM = DPCM) has to be quantized. Quantization takes place by allocating one out of 2^B quantization levels to the quantizer input, where B is the number of bits assigned for the coding of a subband. Next, the quantized value is coded into B bits.
Design of an efficient encoding scheme requires some knowledge of the statistics of the signal to be coded. If we had an a priori knowledge of the statistics of the samples, a nearly optimum quantization scheme would consist of:

- A quantizer matched to the probability density function (PDF) of the samples to be quantized.
- A predictor optimized for the given autocorrelation function of the signal.

In digital speech-encoding systems, we have only a small amount of a priori knowledge of the statistics which, in addition, usually change with time.

- The long-period mean level differs from speaker to speaker.
- At a given mean level, the instaneous level changes because of variations in speech sounds (c.q. split speech sounds).
- The correlations (as far as present) between successive samples change because of variations in speech sounds (c.q. split speech sounds).

To overcome these problems of unknown statistics, adaptive quantization and (in case of DPCM) adaptive prediction schemes can be used.
We will now discuss the PCM coder and decoder design that best suits our purpose. Then the implementation and testing of it is described.

5.2. Design considerations

Also for the coding of the subband signals several methods are possible. Many of them are extensively discussed in [6, 22,23,24,25].
Concerning the quantization, there are two possibilities, fixed and adaptive quantizers. For reasons like unknown mean level and variations of the instantaneous level, a nonadaptive quantizer will not provide an acceptable quality in our situation. Therefore adaptive quantization has been chosen, in which the 2^B possible quantization levels are not fixed in time. For the adaption two schemes are possible. With the first one, forward adaption, the adaption value is calculated from samples of the input signal. As this has to take place in each subband, this method is far too complex and furthermore requires serial buffering (extra delay) and a lot of side information for transmitting the adaption values to the receiver. The second scheme, backward adaption, is more suitable. Here, the adaption value is calculated from quantized samples. In the receiver the same can be done and therefore no side information is required. Furthermore no extra delay above the filtering delay time occurs, since the buffering required is done in parallel. Hence for our SBC design a backward adaptive quantization scheme (AQB) has been chosen.
For the application of DPCM the following has to be taken into account. With DPCM the complexity of the coder algorithms, especially when the prediction is made adaptive, increases. Furthermore, as there is little or no correlation between the subband outputs [2], the predictor coefficients will be close to zero [26]. So DPCM will perform little or no quality improvement, while the complexity increases. For these reasons it has been decided to use "simple" PCM coding without prediction.
We will now discuss the backward adaptive PCM method, chosen for the (de-)coding of the subband signals, in more detail. See also [6,7,15,26].
A general scheme of a PCMAQB coder is depicted in Fig. 18.

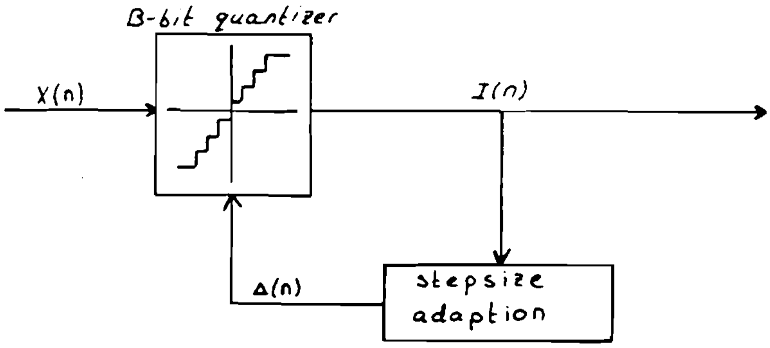


Fig. 18: Block diagram of a PCMAQB coder.

For the quantization of the input signal $x(n)$ a quantizer with a midriser characteristic, as shown in Fig. 19, is used. As can be seen from this figure, a value of $x(n)$ in the range $[I(n)*\Delta(n);(I(n)+1)*\Delta(n)\rangle$ results in a quantized value $\hat{x}(n)=(I(n)+0.5)*\Delta(n)$, where:

$\Delta(n)$=stepsize (i.e. spacing between quantized levels) at moment n.

$I(n)$=two's complement value representing one out of 2^B quantization levels, i.e. the B-bit code word, at moment n.

The output of the quantizer is the $\hat{x}(n)$ representing code word $I(n)$. The stepsize adaption strategy used is based on the one-word stepsize memory approach proposed by Jayant, Flanagan and Cummiskey [25]. The robust stepsize adaption is based on the relation:

$$\Delta(n)=[\Delta(n-1)]^{\gamma}*M(|I(n-1)|) \tag{5}$$

where $\Delta(n)$ is the stepsize used for the encoding at the nth time sample and $\Delta(n-1)$ is the stepsize that was used for the (n-1)th time sample. The value of $\Delta(n-1)$ is raised to a power γ, where $\gamma<1$ is a coder parameter (to be discussed in more detail in 5.3.). It is then multiplied by a (positive) scale factor M(.) which is a function of the previous code word $I(n-1)$ to give the stepsize estimate $\Delta(n)$. In general,

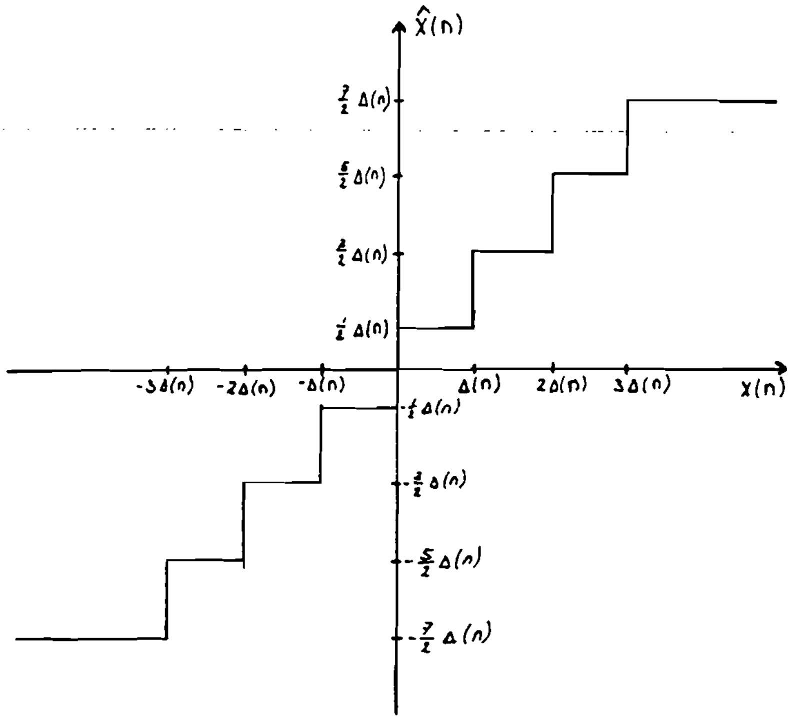


Fig. 19: Uniform quantizer with 8 levels (B=3).

if the previous code word I(n-1) indicates that an upper (absolute) quantizer level was used in encoding, a value of M(.)>1 is used to increase the size of the new stepsize $\Delta(n)$. If I(n-1) indicates that a lower (absolute) amplitude level was used by the quantizer, a value of M(.)<1 is used to reduce the estimation of the new stepsize $\Delta(n)$. Thus the stepsize adaption algorithm is constantly attempting to adjust the stepsize $\Delta(n)$ such that it tracks the rms level of the signal and scales the quantizer characteristic to span the amplitude range of the signal. For practical reasons the stepsize must be limited to the range:

$$\Delta min \leqslant \Delta(n) \leqslant \Delta max \qquad (6)$$

to prevent $\Delta(n)$ to grow beyond the limits of the number representation adopted. The ratio $\Delta max/\Delta min$ determines the dynamic range that the coder can handle. In our case a ratio of 2048, ≈66 dB, is taken, being within the range of the digital arithmetic. The actual values of Δmin and Δmax must be different for each subband (each subband signal is coded with its own PCMAQB coder), to match properly the dynamic range characteristics of the coders to that of the subband signals, as shown in Fig. 15.

The proportion of the amplitude range that is spanned by the quantizer at a particular time (i.e. for a particular stepsize $\Delta(n)$) determines its "loading". If the range of the quantizer is too small relative to the signal range, the quantizer will overload and clip the signal. If it is too large, the quantizer stepsize will be too large, and this will result in an excessive quantization error or noise (often referred to as granular noise). Thus the proper loading of the quantizer is an important factor in maintaining a good reproduction of the signal. The loading is controlled by the choice of the parameters γ and M(.).

In the receiver the same stepsize adaption algorithm is used as in the transmitter, since this adaption takes place according to the coded subband samples. A block diagram of a PCMAQB decoder is depicted in Fig. 20. By consequence of the midriser characteristic of the dequantizer an input I(n) results in the dequantized value $\hat{x}(n)=(I(n)+0.5)*\Delta(n)$.

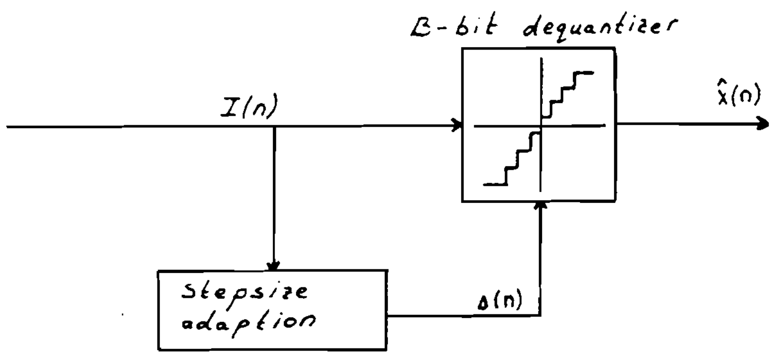


Fig. 20: Block diagram of a PCMAQB decoder.

5.3. Implementation of the chosen (de)coder algorithm

By taking the logarithm, eq. (5) can be written in the form

$$d(n)=\gamma*d(n-1)+m(|I(n-1)|) \qquad (7)$$

where:

$d(n)=\log\Delta(n)$

$m(.)=\log M(.)$

The stepsize adaption is then implemented by the circuit shown in Fig. 21.

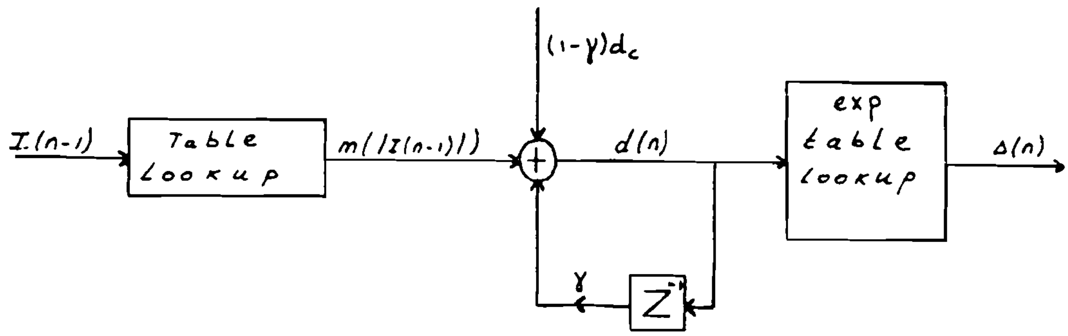


Fig. 21: Stepsize adaption circuit.

The first table lookup converts values of I(n-1) to m(|I(n-1)|), and the second table lookup realizes the exponential conversion from d(n) to Δ(n). However, as for the quantization the value 1/Δ(n) is needed, in the transmitter an exponential conversion from d(n) to 1/Δ(n) has been implemented. Thus it is seen that the adaption circuit consists of two table lookups and a first-order recursive digital filter which can easily be implemented. The advantage of the method of using table lookups is, that it can be done relatively fast and it can perform functions difficult to calculate in the DSP.

An extra dc input $(1-\gamma)*d_c$ is also applied to the circuit in Fig. 21, and it is used to control the loading of the quantizer. Thus eq. (7) has been modified to the form:

$$d(n)=\gamma*d(n-1)+m(|I(n-1)|)+(1-\gamma)*d_c \qquad (8)$$

where for d_c has been chosen the practical value of approximately

$$d_c \approx \log(\Delta max/10) \qquad (9)$$

Now it is the right moment to discuss the parameter γ. The adaption leakage factor γ, chosen to be $\gamma=0.99$, forces the realignment of the stepsizes between transmitter and receiver after channel errors occur. Realignment will also occur when the stepsize reaches its maximum or minimum value according to eq. (6), even if γ were chosen to be 1. However, a long time may pass before this maximum or minimum is achieved. Since the cancellation of aliasing in the QMF bank depends strongly on the exact tracking of the stepsizes in each subband, it is therefore preferable to use a value of $\gamma<1$ to dissipate any effects of channel errors more rapidly. Another effect of the leakage due to γ is that the log of the stepsize, that is d(n) in Fig. 21, tends to decay to zero in the absence of the inputs $m(|I(n-1)|)$ and $(1-\gamma)*d_c$. By adding the term $(1-\gamma)*d_c$ the stepsize toward which d(n) decays can be set to any arbitrary level.

We will now discuss the implementation of the table lookups (stored in program memory) in more detail. In the first table lookup I(n-1) is converted to $m(|I(n-1)|)$ according to:

$$I(n-1)\longrightarrow|I(n-1)|\longrightarrow M(|I(n-1)|)\longrightarrow m(|I(n-1)|) \qquad (10)$$

For each subband the same table is used. An appropriate base for taking the logarithms has been found to be 10^4. Choices for the values of M(.) for different numbers of bits to which the input signal has to be quantized are depicted in Table 4. Experiments have shown that small deviations from the presented values have very little influence on the performance of the adaptive coder. The optimal representation format for the m(.) values is the Q18 format.
The exponential table lookup is not the same for each subband. Each subband has its own dynamic range spanned up by Δmin and Δmax. The ratio $\Delta max/\Delta min$ is the same (2048) for each subband.The values chosen for Δmax and Δmin, which have been adapted to the dynamic ranges of the subband signals, are depicted in Table 5. The six different tables have been

Table 4: PCMAQB coder parameters.

B=	4	3	2
M_1	0.9	0.85	0.85
M_2	0.9	1.0	1.9
M_3	0.9	1.0	
M_4	0.9	1.5	
M_5	1.2		
M_6	1.6		
M_7	2.0		
M_8	2.4		

obtained by taking the logarithm of Δmax and Δmin and dividing the range between dmin and dmax into 127 uniformly spaced parts, resulting in 128 entries (convenient for our purpose). For each entry the exponential conversion to Δ (in the transmitter to 1/Δ) has been stored in the table. The tables, created in this way, can be implemented very efficiently. If each table is represented in its own optimal format, only one table with 128 entries is needed to represent all six tables.

Table 5: Δmin and Δmax for different subbands.

subband	f(Hz)	Δmin	Δmax	representation format for Δ
1	0-500	4.882813E-4	1	Q15
2	500-1000	4.882813E-4	1	Q15
3	1000-1500	2.441406E-4	0.5	Q16
4	1500-2000	2.441406E-4	0.5	Q16
5	2000-2500	3.051758E-5	0.0625	Q19
6	2500-3000	3.051758E-5	0.0625	Q19
7	3000-3500	-	-	-
8	3500-4000	-	-	-

For example, Δmin and Δmax for subband 1 are optimally represented in Q15 format, resulting in the integer values to be implemented of 16 resp. 32767. For subband 5 Δmin and Δmax are optimally represented in Q19 format, which also results in the integer values of 16 resp. 32767. The same holds for intermediate values of Δ. So for each subband this table has to be interpreted with its own representation format and its own input range dmin to dmax. The entries of this table, d(n), are for each subband optimally represented in Q14 format. So this way of implementing allows us to work with six exponential table lookups, matched to the dynamic ranges of the different subband signals, for the price of only one.

Since the six PCMAQB coders, as mentioned above, have a lot in common, they have been all brought together in one PCMAQB coder subroutine, that in some points is passed through differently for the different subband signals. To accomplish this, tables in program memory are used for storing constants as $(1-\gamma)*d_c$, dmin, dmax and the bit-assignments, and for updating variables as d(n) and I(n). These constants and variables are different for each subband and therefore each table contains one entry for each subband. To synchronize with the rest of the SBC program, again the variable PATH is used to determine the subband to be coded. The same holds for the PCMAQB decoders in the receiver.
The essential points in the coding and decoding subroutines, see appendix D resp. G, are discussed below. For a listing of the table lookups is referred to the program memory initializations in appendix B and F.

After the QMF routine, in the transmitter, has delivered a subband sample to the PCMAQB coding routine, the following takes place. First the subband sample is identified using PATH (=0 to 7). This means reading in the number of code bits, dictated by the voicing strategy in use, and reading in the dynamic range (i.e. dmin and dmax) for d(n). Then the stepsize adaption algorithm is executed to determine the new

value of $1/\Delta(n)$ to quantize with. This is done by reading in the previous coded result for that subband, $I(n-1)$, followed by the determination of $m(|I(n-1)|)$, $(1-\gamma)*d_c$ and $\gamma*d(n-1)$. Adding the former three values according to eq. (8) will lead to the new value of $d(n)$. This new value of $d(n)$ is checked on lying within the dynamic range spanned by dmin and dmax; if not, saturation to dmin or dmax takes place. The remaining value of $d(n)$ is saved in its table in program memory. Then $d(n)$ is rounded to one of the entry points of the exponential table lookup, ultimately leading to $1/\Delta(n)$. Having this value of $1/\Delta(n)$ the quantization takes place by multiplying it with the subband sample delivered by the QMF routine and taking the integer part of the product. According to the midriser characteristic of the quantizer this results in the two's complement code word $I(n)$. After checking the range of it, dictated by the number of code bits, and possible saturation the code word $I(n)$ is saved in its table in program memory. Later the multiplexer, to be discussed in chapter 6, will use this table of code words $I(n)$ to create a 16 kbit/s data stream.
In the receiver, the demultiplexer will recover the table of code words $I(n)$ from the 16 kbit/s data stream. This table will be used by the PCMAQB decoder to reconstruct the uncoded subband signals. When this decoding routine is called, first the stepsize $\Delta(n)$ is determined, in the same way as described for the transmitter. Then the subband sample to be decoded, according to PATH, is read in. After adding 0.5 to this value of $I(n)$, multiplication with $\Delta(n)$ takes place, resulting in a dequantized subband sample according to the midriser characteristic. After checking the range of this reconstructed subband sample and possible saturation, it is delivered to the inverse QMF bank for further processing. For more details about the coding and decoding algorithm is referred to the source files, listed in appendix D and G.

5.4. Tests and results

For testing the implemented PCMAQB coders and decoders, again the test configuration as shown in Fig. 13 has been used. Only the DSP programs used in this stage differ from those used in Fig. 13. The main routine in the transmitter DSP has been extended with the PCMAQB coding module. After the QMF program has finished, the PCMAQB program is passed-through. Then the wait cycle follows. The same holds for the main routine in the receiver DSP. However, here the PCMAQB decoding is passed-through before the inverse QMF program is called. The configuration of the DSP programs, also including the extensions for the semi-adaptive bit-allocation, is sketched in Fig. 22.

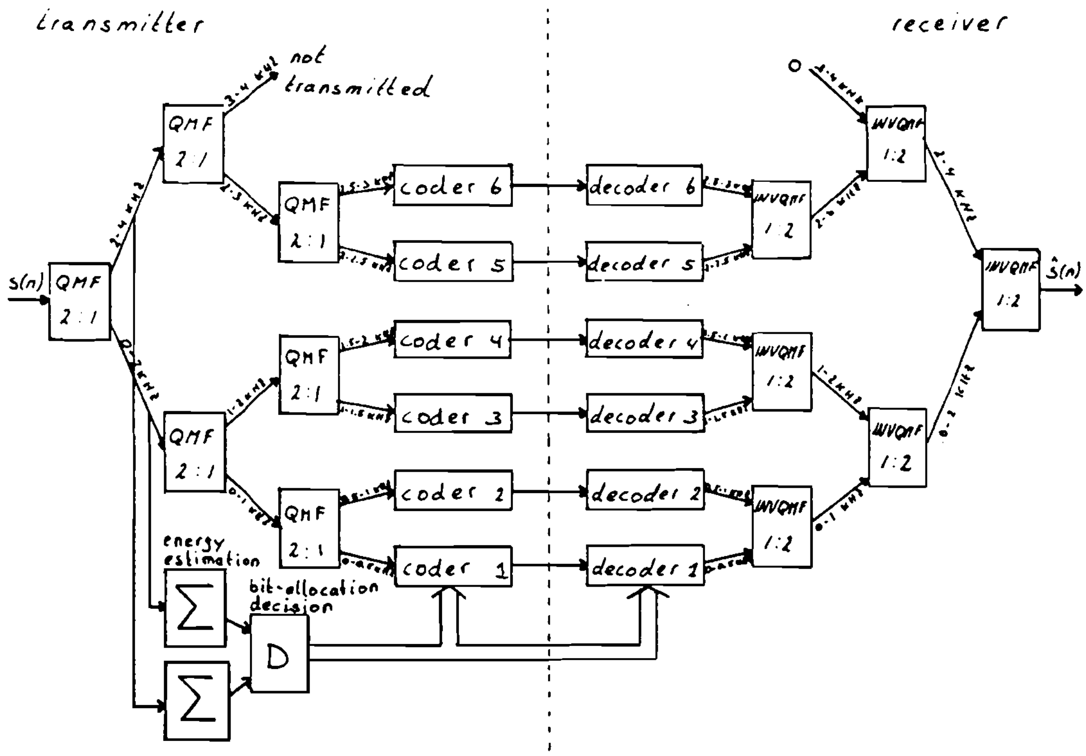


Fig. 22: DSP configuration for testing the PCMAQB coders and decoders.

Again tests with speech signals have been carried out, which is necessary due to the adaptive schemes in the SBC algorithm. In all tests the analog speech input was applied to the transmitter system input, and listening to the reconstructed speech signal took place via the audio output of the receiver system. For some kind of objective checking, the reconstructed speech signal was also visualized on a memory oscilloscope. Furthermore, the reconstructed speech quality could easily be compared with the maximum quality that can be achieved (up to 3 kHz bandlimited speech quality, see 4.4.) by skipping the PCMAQB coding and decoding (by means of a hardware switch connected with the DSP's).

5.4.1. Performance of the coders and decoders

To confirm the proper working of the coding and decoding itself, the influence of the bit-allocation assignments was eliminated by coding each subband signal with 4 bits. This was allowed, by the fact that the communication between transmitter and receiver system still was performed in a parallel way and not serial. The performance of this 24 kbit/s SBC was rather good. The reconstructed speech differed only very little when compared with uncoded, up to 3 kHz bandlimited, speech.
The loading of the (de-)quantizers was subjectively checked by switching off (in software) the saturation actions in the coding and decoding, since many saturation actions indicate an improper loading of the (de-)quantizers. This resulted in some quality degradation, but not so much as to conclude that the (de-)quantizers were improperly loaded.
Also, the proper loading of the (de-)quantizers has been confirmed objectively, as the behaviour of the reconstructed speech signal showed no excessive clipping on the memory oscilloscope. Finally, also the correctness of the code words in the tables of the transmitter and receiver programs was verified.

After the proper working of the coder and decoder principle had been confirmed in this way, the coding into 16 kbit/s, i.e. the influence of a bit-allocation different from 4 bits per subband sample, could be tested.

5.4.2. Performance of the chosen bit-allocation method

First the semi-adaptive bit-allocation method with the assignments as shown in Table 3, (eliminated in the former testing) was restored. Then the same tests, as described in 5.4.1., have been carried out. The intelligibility of the reconstructed speech was fairly satisfactory. However, a quality degradation, compared to uncoded up to 3 kHz bandlimited speech, was noticeable. Furthermore, tests with several listeners have been carried out to compare the semi-adaptive bit-allocation with fixed-bit-allocation and also to try out other bit assignments. Notwithstanding the fact, that in some cases differences were difficult to perceive, the semi-adaptive bit-allocation with the assignments from Table 3 appeared to provide the best overall performance. However, the difference with fixed-bit-allocation was less than expected from the argument given in 4.2.

Finally the loading of the (de-)quantizers has been investigated, which proved to be proper. This is not only a consequence of a convenient stepsize adaption algorithm but also due to an appropriate representation of the subband signals (see 3.5.) and the use of "six different" exponential table lookups (5.3.). In the development work, preceding the ultimate realization of the coding and decoding described above, also tests have been carried out, where all subband signals were represented in the same Q15 format and were coded using the same exponential table lookup for each subband signal. This resulted in a reconstructed speech signal that excessively suffered from audible clipping and "clicks" due to an improper loading of the (de-)quantizers. Also a dynamic range for the exponential table lookups smaller than 66 dB has been tried out, resulting again in an improper loading of the (de-)quantizers. By the way, due to this dynamic range, the different values of Δmin are such, that in periods of silence ($I(n)=0$ and $\Delta(n)=\Delta min$; reconstructed subband sample $=(0+0.5)*\Delta min$) 1/2 Δmin causes no audible output upon reconstruction.

5.5. Conclusion

The 16 kbit/s coding and decoding of the subband samples, using 2,3 or 4 bit PCMAQB and a semi-adaptive bit-allocation algorithm, results in adding an amount of quantization noise to the replica of s(n) (3.6.). However, the perceived quality of the reconstructed speech signal, especially the intelligibility, is considered acceptable for our applications.
At this stage the actual Subband Coding and Decoding has been realized. The only thing left to be described is the realization of the multiplexing and demultiplexing to perform the serial 16 kbit/s communication.

6. MULTIPLEXING AND DEMULTIPLEXING OF THE CODED SUBBAND SAMPLES

6.1. Introduction

To create, after quantization and coding, the ultimate serial 16 kbit/s data stream, the subband signals together with the side information and synchronization bits are multiplexed. This multiplexing has to be performed in a controlled way; by demultiplexing it has to be possible to recover the coded subband samples, the side information and the synchronization bits. This can be accomplished by multiplexing the data to be transmitted into a repetitive framed sequence. Important features, coupled with the multiplexing and demultiplexing are:

-Synchronization between transmitter and receiver, i.e. frame alignment.

i) The time for alignment with 99% probability must not seriously disrupt the speech communication. It is usually a compromise between the time taken to confirm the presence of the framing pattern and the risk of incorrectly aligning to random imitations of it.

ii) False indications of lost frame alignment for an error rate of ca. $1:10^3$ must not occur too frequently.

-Frame organization.

The composition of a frame in the 16 kbit/s serial data stream has to match the framing, i.e. the subband processing sequences and the voicing strategy changes (Fig. 17) in the "actual" (without (de)multiplexing) Subband Coder and Decoder algorithms. This to avoid excessive buffering.

The bit synchronization in the demultiplexer is assumed to be located in a (higher order) part outside the Subband Decoder algorithm, where the demultiplexing of the Subband Coder signal (16 kbit/s) and another signal (see 1.) has to be done. Therefore the bit synchronization will not be described here.

In this chapter the design, implementation and testing of the multiplexer and demultiplexer for our SBC will be discussed.

6.2. Design considerations

Based upon recommendations from literature [3,27] and previous work [6,8,28] it has been decided to use the following strategies for the framing and synchronization.
To acquire frame alignment, when the first bits are received or alignment has been lost, it is for the demultiplexer to search for and recognize the Frame Alignment Word (FAW), present in a fixed position in each frame. Then the demultiplexer has to lock its timing counters into the correct phase relationship with the incoming signal, and examine the FAW in two successive confirmatory frames. Non-recognition of the FAW in its expected position in either of these two frames causes a recommencement of the search. This is done to safeguard against false alignment by imitations of the FAW within the signal.
The minimum time that is necessary to acquire and confirm frame alignment is, therefore, between 2 and 3 frame periods, depending on the point within the frame that a valid signal is applied and the search commences. However, the incoming signal contains an essentially random occurence of ones and zeros in the information digit time-slots, and there is a probability of these digits imitating the FAW. The probability of an imitation in any position is 2^{-m} for a random bit stream, m being the number of bits in the FAW. Such imitations of the FAW will lead to an increase of the time required to acquire frame alignment, due to the greater number of false starts; the false alignment being rejected at the first or second confirmatory frame.
The time, t_1, to acquire frame alignment with a 99% probability of not being exceeded can be estimated using the following simplified formula for the alignment strategy described above:

$$t_1 = F(3+(F-1)2^{-m}+2*3[(F-1)2^{-m}]^{0.5})/16 \quad \text{milliseconds} \qquad (11)$$

where:

F is the total number of bits in a frame of the multiplexed signal.

m is the number of bits in the FAW.

To avoid false alarms due to bit errors as much as possible, frame alignment is considered to have been lost when 4 consecutive FAW's are incorrectly received in their predicted positions. The probability of random digital errors causing this condition to be fulfilled is approximately $(mp)^4$ when p, the bit error rate, is low; better than, say, $1{:}10^3$. It follows that the mean time between false losses of frame alignment for a given error rate is,

$$t_2 = F/(16(mp)^4) \quad \text{milliseconds} \qquad (12)$$

To meet the requirements i) and ii) in 6.1., t_1 has to be small (order of ms) and t_2 has to be large (order of hours). Appropriate values for the frame length F and the FAW length m have been chosen to be 128 resp. 8 bits. In that case the frame duration is 8 ms and the information bit rate is 15 bits per ms. These choices have already been used in designing the bit-allocation and synchronization for the "actual" Subband Coding and Decoding (chapter 4).

With respect to the organization of a frame, the following can be said. As mentioned above, for synchronization reasons the first 8 bits of such an 128 bits frame contain a FAW. As the side information, required for our SBC design, only concerns the bit-allocation (3 possibilities) used for the coding of the subband samples during a whole frame period, this is inserted in the FAW. So three FAW's are possible, each being composed in a special way to reduce the probability of imitating it:

FAW1 = >76 voiced

FAW2 = >96 intermediate

FAW3 = >CC unvoiced

The remaining 120 bits are all used for transmission of the coded subband samples. These 120 bits are filled in, taking into account the way of subband processing in the "actual" SBC, see Fig. 17.

6.3. Implementation of the chosen multiplexer and demultiplexer algorithm

Before describing the implementation of the multiplexer and demultiplexer themselves, we will discuss the application of the 16 kHz clock signal, needed to create a 16 kbit/s bit stream. As discussed earlier, the "actual" SBC algorithms are controlled by an 8 kHz interrupt signal, updating the timing variables PATH and FRAME. Therefore, to complete the overall SBC system, the DSP programs have been adapted as follows.

The transmitter DSP is controlled by interrupts, from now on a 16 kHz clock signal. Each 16 kHz interrupt created by the A/D converter (see chapter 7) causes an execution of the interrupt service routine. In this interrupt service routine a sample of s(n) is read in, followed by a call of the multiplex subroutine to create one bit to be written out. The main routine is passed-through for every other 16 kHz interrupt (let say the odd ones), realizing the same 8 kHz control as before. In this main routine, thus once interrupted by a 16 kHz interrupt, the QMF filtering and PCMAQB coding is carried out, followed by a wait cycle for the next odd interrupt.

The receiver DSP program is controlled by 16 kHz interrupts from the regained system clock, for which again the transmitter system clock has been used. In the interrupt service routine the demultiplex subroutine is called to read in one bit from the 16 kbit/s stream and furthermore a reconstructed speech sample, ŝ(n), is written out. In the main routi-

ne, to be passed-through for every other interrupt, the PCMAQB decoding and the QMF reconstruction takes place, followed by a wait cycle for the next odd interrupt. However, when the receiver is not in alignment $\hat{s}(n)$ is set to zero, without calling the PCMAQB decoding and QMF reconstruction routines.
Furthermore, considering the 16 kHz transmitter input, $s(n)$, only one out of two inputs is used in the main routine. In the receiver, the reconstructed output from the inverse QMF filtering is a sample sequence with a rate of 8 kHz. To reduce the aperture effect [14, pp. 302-304] this sequence is interpolated to obtain the ultimate output signal, $\hat{s}(n)$, with a sampling rate of 16 kHz.

To accomplish the whole synchronization, beside the variables PATH and FRAME a variable called STATE is used. STATE is a modulo-16 counter, being updated for every 16 kHz interrupt. So, for every odd value of STATE, PATH is updated and the main routine (in transmitter or receiver) is passed-through. As discussed in 4.3., FRAME is updated for every sequence of PATH from 0 to 7, which has to be done eight times for one frame period in the "actual" Subband Coding or Decoding.
Now the multiplexer and demultiplexer implementations are discussed in more detail.

6.3.1. Multiplexer

Apart from timing considerations, the multiplex algorithm is not very complicated. Each time the multiplex routine is called, a variable is checked to determine the sort of information, i.e. from the FAW or a certain subband, to be transmitted. A second variable is checked to determine which bit from that FAW or subband has to be transmitted. The transmission of each bit is done via a send buffer. Each time before the first bit of a FAW or coded subband sample has to be transmitted, the send buffer is loaded with that FAW or subband sample code word. The FAW is retained from the bit-allocation decision network, while the coded subband samples are retained from the "I(n) table" (see 5.3.) in program memory. The variable that indicates the bit to be transmitted is updated for each bit sent. The variable that indicates the sort of information to be transmitted is updated after the last bit of the send buffer has been sent. To match the way of subband processing in the QMF splitting and PCMAQB coding (Fig. 17) first the bits from subband 1 are sent, followed by sending the bits from resp. subband 4,5,2,3 and 6. This sequence is repeated eight times for one frame. The sending of these eight sequences is preceded by sending the FAW of the frame. Furthermore, to ensure that each subband sample is retained from the "I(n) table" on the right moment (i.e. after it has been created and before the next sample of that subband is created) the multiplexer frame periods are delayed three 16 kHz interrupts (being the minimum possible) with respect to the frame periods in the "actual" transmitter SBC. The subband processing in the QMF splitting and PCMAQB coding for one frame period, and its corresponding multiplexer frame for the three voicing strategies resp. voiced, intermediate and unvoiced, are depicted in Fig. 23. In this figure, the numbers in the upper line represent the subband processed in the QMF splitting and PCMAQB coding between two odd 16 kHz interrupt service routines. The numbers in the next line represent the subband

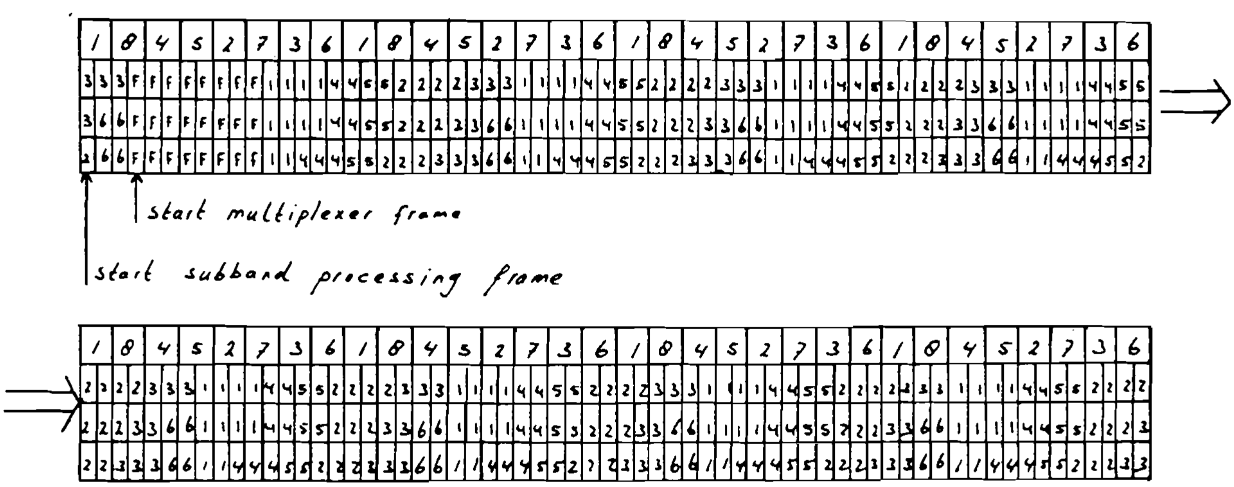


Fig. 23: Frame composition in multiplexer with respect to a subband processing frame.

from which a bit is sent in a 16 kHz interrupt service routine for the voiced bit-allocation strategy; a "F" represents a FAW bit. The same holds for the third and fourth line, but then for the intermediate resp. unvoiced bit-allocation strategy.
The source file of the multiplex subroutine is listed in appendix E.

6.3.2. Demultiplexer

The demultiplex algorithm is more complex, as it also has to take care of the frame alignment. Each time the demultiplex routine is called, it is determined whether the receiver is in alignment or not.
When the receiver is not in alignment, either a search for a FAW takes place or, when a FAW already has been detected, the confirmatory stage is executed. For the FAW search, a received bit is inserted in the receive buffer (a shift register), which is subsequently checked on containing a FAW. If a FAW is detected, then in the next demultiplex call jumping to the confirmatory stage takes place, otherwise again a FAW search is executed. In the confirmatory stage, a received bit is inserted in the receive buffer and, if it is the moment to expect a FAW (128 bits after the previous FAW), the receive buffer is checked on containing a FAW. In the demultiplex calls, jumping to this confirmatory stage repeats until alignment is definitively confirmed or denied. When denied, the next demultiplex call concerns a FAW search. When confirmed, the receiver timing is set by loading the synchronization variables STATE, PATH and FRAME with the appropriate values, and in the next demultiplex call demultiplexing takes place according to the "in alignment situation".
When the receiver is in alignment, the inverse of the multiplex algorithm is executed. Again two variables are used to determine which bit for which subband or FAW is to be received. The received subband samples are stored in the "I(n) table" (see 5.3.) in program memory. The side information

retained from a received FAW is delivered to the voicing strategy buffer. The only action without a corresponding action in the multiplexer, is the alignment check when a FAW is received. When 4 consecutive FAW's are incorrectly received in their predicted positions, the "in alignment situation" is left and the next demultiplex call results in a FAW search.

The demultiplexer frame for the three possible voicing strategies, matching the subband processing in the PCMAQB decoding and QMF reconstruction for a corresponding frame period, is depicted in Fig. 24. As the subband samples first

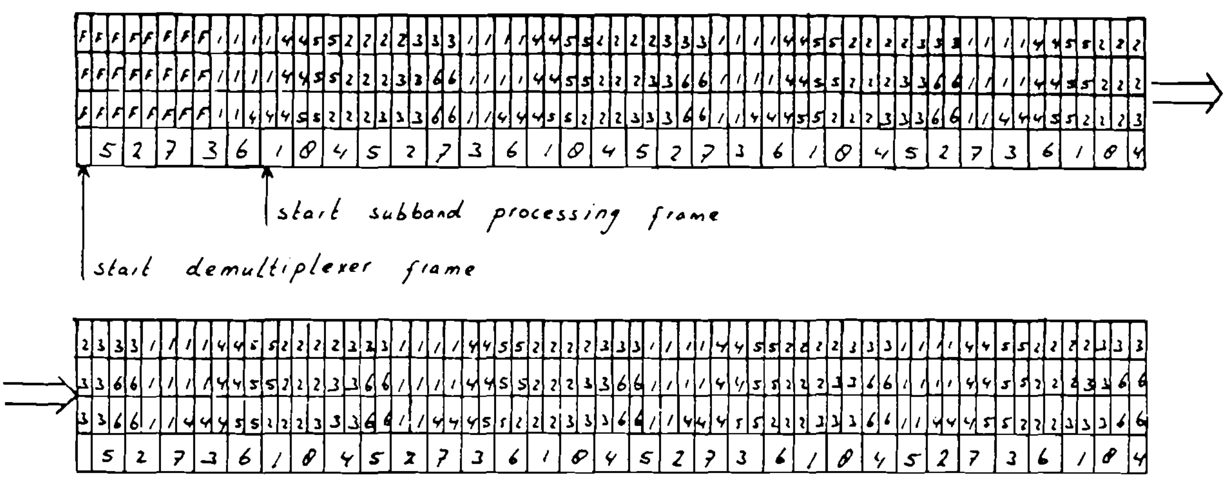


Fig. 24: Frame composition in demultiplexer with respect to a subband processing frame.

have to be received before they can be processed, in this figure the numbers in the upper three lines represent the subband or FAW (F) for which a bit is received for the resp. bit-allocation strategies: voiced, intermediate and unvoiced. The numbers in the last line represent the subband processed in the PCMAQB decoding and QMF reconstruction ("actual" Subband Decoding). As can be seen from this figure, to process each subband sample (retained from the "I(n) table") on the right moment, the frame periods in the "actual" Subband Decoding are delayed eleven 16 kHz interrupts (being the minimum possible) with respect to the demultiplexer frame periods.

The source file of the demultiplex subroutine is listed in appendix I.

6.4. Experimental confirmation of the proper working of the multiplexer and demultiplexer

With the configuration from Fig. 13, the 8 kHz A/D and D/A conversions replaced by 16 kHz A/D and D/A conversions and the DSP programs updated to perform the functioning depicted in Fig. 25, experiments have been carried out. Again analog speech was applied to the transmitter system input and the audio output of the receiver system was used to listen to the reconstructed speech.

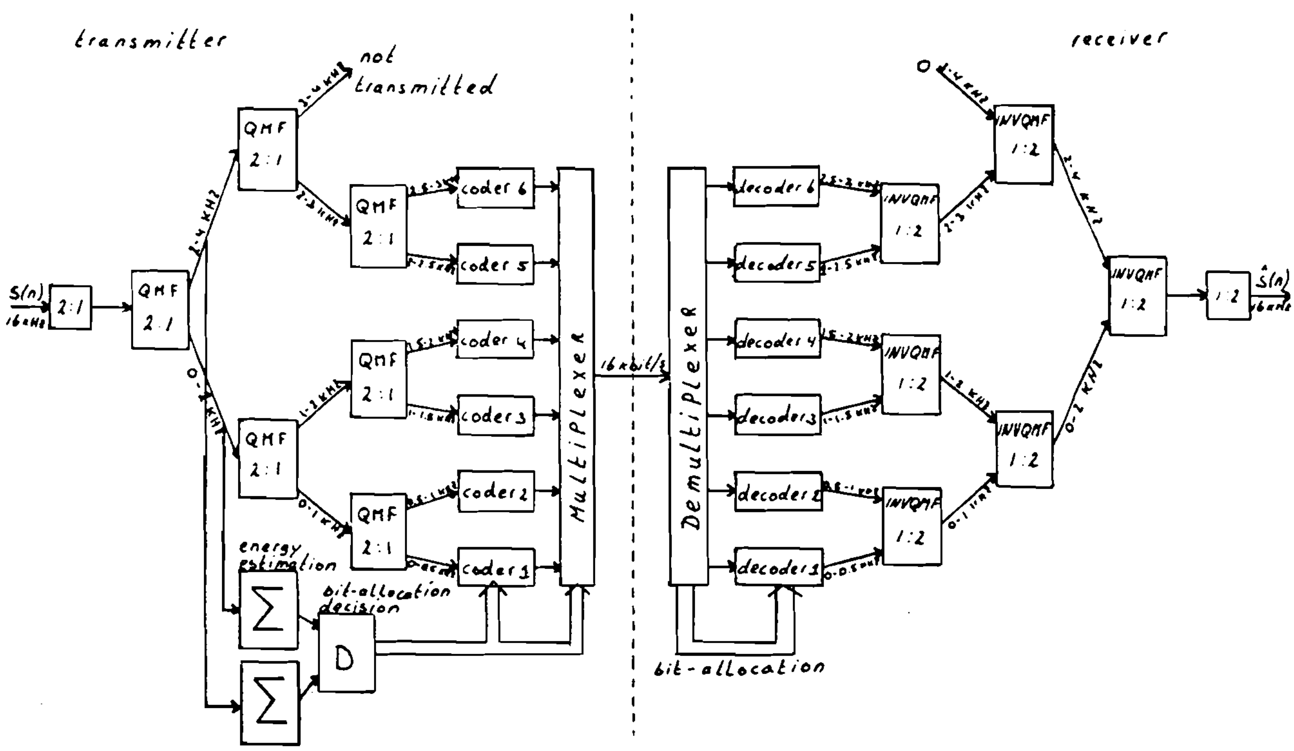


Fig. 25: Block diagram of the implemented Subband Coder and Decoder on two DSP's.

After starting up the transmitter and receiver program, frame alignment in the receiver was acquired very fast (hardly to perceive). When in alignment, the perceived quality obtained with this "serial 16 kbit/s SBC" was exactly the same as that obtained for the "parallel 16 kbit/s SBC" (5.4.2.). Furthermore, proper functioning in situations that cause loss of frame alignment and frame realignment has been

confirmed by imitating these situations (interrupting the communications and/or synchronization).
Also, for these situations, the necessity of an adaption leakage factor γ<1 in the PCMAQB (de-)coding, eq. (8), has been proved. Choosing this factor equal to 1 resulted in a lasting mis-adaption between the PCMAQB coding and decoding stepsizes, leading to unintelligible reconstructed speech.

So, also the multiplexer and demultiplexer implementations may be considered to function properly. With that, as these were the last modules to be added, the description of the SBC realization concerning the DSP implementations has finished. Therefore Fig. 25 is the block diagram of the implemented Subband Coder and Decoder.
All the necessary source files are listed in appendix B to E for the transmitter DSP, and in appendix F to I for the receiver DSP.

7. HARDWARE CONSIDERATIONS

7.1. Introduction

So far the hardware environment of the DSP's, also necessary for a proper working SBC realization, has been left undiscussed.
In this chapter we will discuss the essentials of the bandlimiting, the A/D and D/A conversion and furthermore the serial transmission from transmitter to receiver DSP.
For detailed information about the A/D and D/A converters, and for a hardware scheme of the AIB (pin connections etc.) is referred to the AIB book [11].

7.2. I/O of the transmitter DSP

The 16 kHz 12-bit linear PCM digital input, s(n), to the transmitter DSP, and the 16 kHz interrupt signal that controls the DSP program (6.3.) are created in the following way.
The analog speech signal, s(t), is bandlimited, for which in our case a standard telephone-channel filter (appendix A) has been used. Then the signal is applied to J2 of the AIB, which is via a sample-and-hold-circuit the A/D converter input. Via the software in the DSP program, this A/D converter is programmed to operate with a sampling rate of 16 kHz, and to deliver beside the s(n) signal also a 16 kHz interrupt signal to the DSP. In short the A/D converter works as follows. Upon receipt of a start-of-conversion signal (programmed to be a 16 kHz clock signal) the A/D conversion starts, and after the conversion has been completed, an end-of-conversion signal is created and applied to the interrupt pin of the transmitter DSP. This causes an interrupt (16 kHz) to read in the A/D converted sample and to control the DSP program. The end-of-conversion signal is also used for taking a new sample in the sample-and-hold-circuit. Furthermore, the analog speech signal is supposed to match the dynamic range of the A/D converter, which can be achieved by

means of an amplifier.
The serial 16 kbit/s output stream from the transmitter can be retained from pin 38 of port P1 (a buffered AIB output port). This port P1 is programmed via the DSP program to operate in the sample delay mode, to ensure periodicity in the 16 kbit/s output stream. This means that an output bit from the transmitter DSP is first stored in a primary buffer. By means of a pulse from the same 16 kHz start-of-conversion signal as mentioned above, it is transferred to a secondary buffer, being the interface with the "outside world". As the primary buffer is filled via the DSP program far before a start-of-conversion pulse occurs, it always contains stable data on the moment of transfer.

7.3. I/O of the receiver DSP

In the receiver system, the clock signal that controls the DSP program is not created on the AIB itself as is the case for the transmitter system. In the receiver system an externally regained 16 kHz system clock (to be discussed in 7.4.) controls the synchronization.
The serial 16 kbit/s input stream to the receiver system has to be delivered to pin 12 of port P1. Via the DSP program this AIB input port is programmed to operate in the asynchronous receive mode. By means of a pulse from the regained system clock a bit from the input stream is clocked into an input buffer. As the regained system clock is applied to the interrupt pin of the DSP, this pulse also creates an interrupt to read in the buffered input bit and to control the DSP program.
The 16 kHz digital output samples, $\hat{s}(n)$, from the receiver DSP are delivered to the D/A converter that is programmed to operate in the transparent mode. Each sample is directly D/A converted without double buffering. Next, the D/A converted signal is bandlimited with a up to 4.7 kHz bandlimiting filter resident on the AIB. Finally, the analog output is available via J3, or, when an audio output is desired, via J4 (AIB outputs).

7.4. Communication between transmitter and receiver system

As mentioned in 6.1., in the receiver the system clock is assumed to be regained in a (higher order) part outside the subband decoder algorithm. Therefore, for our (testing) purposes this clock signal has been retained from the transmitter system (hard-wired clock). The signal taken, is the 16 kHz end-of-conversion signal. An end-of-conversion pulse occurs ca. 25 μs after a start-of-conversion pulse used in the transmitter system to clock-out a bit of the 16 kbit/s stream. So, when this end-of-conversion signal is used as a regained system clock in the receiver, always stable data are clocked into the input buffer of the receiver system. Furthermore in this way the transmitter and receiver DSP programs are controlled by the same 16 kHz clock.
In the transmitter system, the end-of-conversion signal is retained from pin 9 of U40 on the AIB, and is via a 50 Ω line driver (SN74S140N) applied to a BNC connector. In the receiver system, this end-of-conversion signal is via a BNC connector applied to pin 13 of port P1.
If both pin 38 of port P1 in the transmitter system and pin 12 of port P1 in the receiver system have been connected with a BNC connector too, the communication between transmitter and receiver system is realized by means of two coaxial cables. One for the 16 kHz system clock and one for the 16 kbit/s serial data stream.
With a frequency counter the system clock frequency has been measured to be 15.974 kHz.

7.5. Outline of the hardware configuration that realizes a 16 kbit/s Subband Coder and Decoder

In Fig. 26 a simple diagram of the SBC hardware and its connections is shown. Furthermore, for the AIB's, to perform the functioning as described in this chapter its jumpers have to be set according to the settings depicted in Table 6.

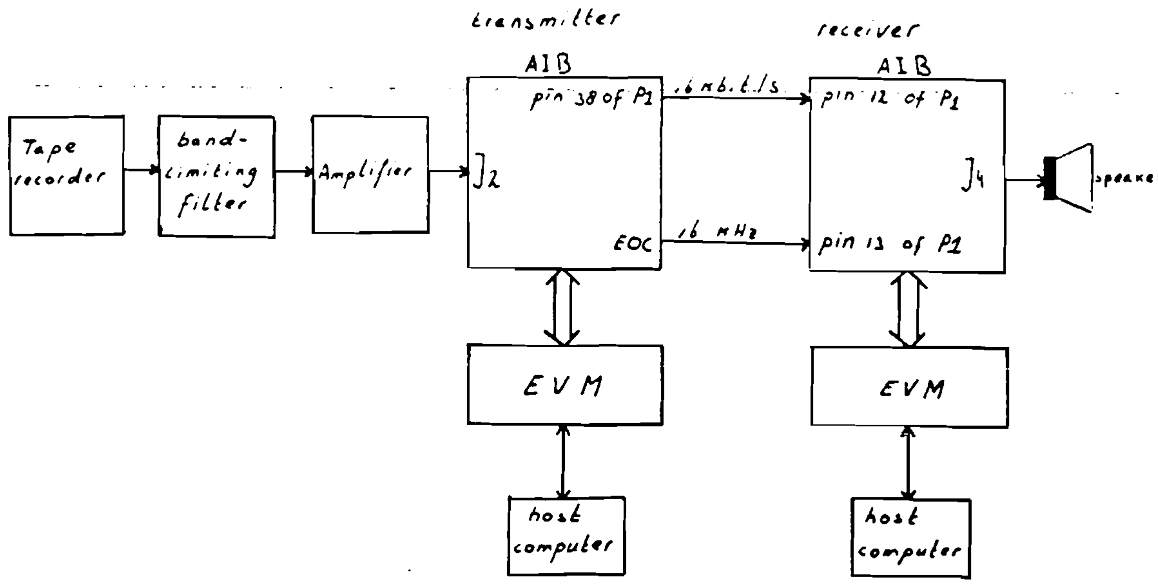


Fig. 26: SBC hardware configuration.

Table 6: AIB jumper settings.

jumper	setting	description
		transmitter
E1	4-5	Connects J2 input jack to A/D (bypasses filter)
E2	don't care	
E3	don't care	
E4	not connected	Leaves Vcc pin on target socket open
E5	2-3	Connects A/D end-of-conversion signal to interrupt pin of TMS32010 emulator socket
E6	1-2	Connects sample and hold to A/D input
		receiver
E1	don't care	
E2	1-2	Connects output filter to J3 output jack
	3-4	Connects D/A converter to output filter
E3	1-2	Connects analog output to audio amplifier
E4	not connected	Leaves Vcc pin on target socket open
E5	2-3	Connects regained system clock to interrupt pin of TMS32010 emulator socket
E6	don't care	

8. UTILIZATION OF THE TRANSMITTER AND RECEIVER DSP

At all stages in the design of the Subband Coder, a prime consideration was the optimum use of the DSP resources. Table 7 illustrates the allocation of processing time for a 0.125 ms period (in which one 8 kHz speech sample is processed) and the utilization of program and data memory. The brackets () indicate the percentage use of the total available resource. Concerning the processing time, the worse case situations are depicted. Table 8 shows the decoder utilization for the inverse processes to restore the original speech signal.

Table 7: Transmitter DSP utilization

function	processing time instruction cycles/0.125 ms	program memory RAM locations
Initialization	-	391
Main program	19	16
Interrupt service routine	68	34
QMF subroutine	217	668
PCMAQB subroutine	110	110
Multiplex subroutine	80	282
total	494 (79%)	1501 (37%)

Data memory (number of RAM locations)=125 (87%)

Table 8: Receiver DSP utilization

function	processing time instruction cycles/0.125 ms	program memory RAM locations
Initialization	-	391
Main program	22	23
Interrupt service routine	74	40
Inverse QMF subroutine	230	675
PCMAQB subroutine	117	118
Demultiplex subroutine	105	396
total	548 (88%)	1643 (40%)

Data memory (number of RAM locations)=129 (90%)

Considering processor speed and data RAM utilization, we may conclude that the DSP capacities are almost completely exploited. Using one DSP for the transmitter and one for the receiver, a more complex SBC algorithm can most probably not be realized.

9. SUGGESTIONS FOR (POSSIBLE) IMPROVEMENT OF THE SUBBAND CODER PERFORMANCE

So far, the bandlimiting in the transmitter and receiver occurs via resp. a standard telephone-channel filter and a lowpass filter resident on the AIB. The performance of the SBC will improve when bandlimiting filters, matching the band of interest 200-3200 Hz (4.2.) more properly, are used. These bandlimiting filters can be realized in an analog or digital form. During previous work [7, pp. 45-58] such a digital bandlimiting filter has been developed. However, adding this algorithm to our SBC program will largely exceed the DSP capacity for both transmitter and receiver. And therefore the application of digital bandlimiting filters will require another two DSP's, one for the transmitter and one for the receiver.

One of the reasons for not choosing DPCM for the coding of the subband signals, was the fact that little or no correlation exists between the subband samples. However, according to [29] some correlation exists in the first two subbands (0-500 Hz and 500-1000 Hz). In coding, whitening these two subband signals using DPCM in stead of PCM will probably improve the SBC performance. It is questionable, however, whether the DSP's can cope with the resulting increase of complexity.

Besides the literature referred to so far, also the references [30] to [60] have been studied. Here, some alternative SBC algorithms are presented. However, our SBC implementation still remains the best compromise between quality and complexity for our purposes.

10. CONCLUSIONS

The Subband Coding technique, using QMF filters for the bandsplitting and reconstruction, PCMAQB for the coding and decoding of the subband signals and furthermore applying a semi-adaptive bit-allocation strategy, has proved to be a very appropriate method for the realization of a 16 kbit/s speech coder and decoder on one DSP each.
Reviewing the design constraints, the quality performance of the implemented SBC, especially the intelligibility of the reconstructed speech, is fairly satisfactory. The capacities of the DSP's are almost completely utilized. So, also the TMS32010, selected for implementing the SBC algorithm, has showed to be a suitable choice.

For professional applications (i.e. without development systems) in future, the Subband Coder will consist of the following main parts:

- a digital or analog bandlimiting filter (200-3200 Hz).
- an amplifier to accomplish the matching of the dynamic ranges of the input speech signal and the A/D converter.
- a 16 kHz sample and hold circuit.
- a 16 kHz 12 bit linear PCM A/D converter.
- a TMS32010 DSP; program memory is partly present in on-chip or off-chip PROM (burnt-in with the Subband Coder program) and partly in off-chip RAM (for tables and variables, not fixed in time).
- a power supply.

The main parts of the Subband Decoder will be:

- a TMS32010 DSP; program memory is partly present in on-chip or off-chip PROM (burnt-in with the Subband Decoder program) and partly in off-chip RAM (for tables and variables, not fixed in time).
- a 16 kHz 12 bit linear PCM D/A converter.
- a digital or analog bandlimiting filter (200-3200 Hz).

-an audio amplifier+loudspeaker.
-a power supply.

Since both Subband Coder and Decoder will be part of a satellite communication system (see 1.), their 16 kHz system clocks can be retained from the (64 kHz) master clock available in the transmitter and receiver of the satellite link.

REFERENCES

[1] A.P. Verlijsdonk and S.H. Mneney,
"Audio-Visual broadcast to rural areas over narrowband satellite channels".
Proceedings of the Africon 1983, pp. B.2.2.1-B.2.2.6, 7-9 December 1983.

[2] F.A. Westall and R.B. Hanes,
"Efficient realizations of digital speech coders for telecommunications applications",
Proc. Int. Zurich Sem., pp. A.2.1- A.2.7, 1984.

[3] R.B. Hanes, F.A. Westall and C. Goody,
"A 16 kbit/s speech coder using a single DSP device",
Proc. IEEE ICASSP, pp. 27.12.1-27.12.4, May 1984.

[4] R.B. Hanes,
"A 16 kbit/s speech codec for four-channel 64 kbit/s transmission",
B.S.T.J., vol. 3, no. 1, pp. 5-13, Jan. 1985.

[5] R. Montagna, L. Nebbia, G. Pirani, F. Rusina,
"A 16 kbit/s speech codec on a single chip DSP",
CSELT Technical reports, vol. 12, no. 4, pp. 409-412, Aug. 1984.

[6] R.P.J. Kleuters,
"Subband Codering",
Project Report, Telecommunications Division of the Eindhoven University of Technology, the Netherlands, 14 Nov. 1984.

[7] J.H. Bolt,
"Speech coding into 16 kbit/s with subband coding on a digital signal processor",
Graduation Work Report, Telecommunications division of the Eindhoven University of Technology, the Netherlands 25 June 1986.

[8] P.J.J. Chitamu,
Practical Training Report, Telecommunications Division of the Eindhoven University of Technology, the Netherlands, July 1986.

[9] TMS32010 User's Guide,
Digital Signal Processor Products, Texas Instruments, SPRU001A, USA 1983.

[10] TMS32010 Evaluation Module User's Guide,
Digital Signal Processor Products, Texas Instruments, SPRU005A, USA 1985.

[11] TMS32010 Analog Interface Board User's Guide,
Digital Signal Processor Products, Texas Instruments, USA 1984.

[12] TMS32010 Assembly Language Programmer's Guide,
Digital Signal Processor Products, Texas Instruments, SPR002B, USA 1985.

[13] TMS32010 Crossware Installation Guide,
Digital Signal Processor Products, Texas Instruments, SPDU049, USA 1986.

[14] A.B. Carlson,
Communication Systems, second edition,
Tokyo: McGraw-Hill Kogakusha, 1975.

[15] R.E. Crochiere,
"On the design of Sub-band Coders for low-bit-rate speech communication",
B.S.T.J., vol. 56, pp. 747-770, May-June 1977.

[16] R.E. Crochiere,
"Digital coding of speech in sub-bands",
Proc. IEEE ICASSP, pp. 233-236, 1976.

[17] R.E. Crochiere, S.A. Webber and J.L. Flanagan,
"Digital coding of speech in sub-bands",
B.S.T.J., vol. 55, pp. 1069-1085, Oct. 1976.

[18] A.J. Barabell and R.E. Crochiere,
"Sub-band coder design incorporating quadrature filters and pitch prediction",
Proc. IEEE ICASSP, pp. 530-533, July 1979.

[19] D. Esteban and C. Galand,
"Application of quadrature mirror filters to split band voice coding schemes",
Proc. IEEE ICASSP, pp. 191-195, May 1977.

[20] R.E. Crochiere,
"Digital signal processor: Sub-band coding",
B.S.T.J., vol. 60, pp. 1633-1653, Sept. 1981.

[21] J.D. Johnston,
"A filter family designed for use in quadrature mirror filter banks",
Proc. IEEE ICASSP, pp. 291-294, April 1980.

[22] V. Gupta and K. Virupaksha,
"Performance evaluation of adaptive quantizers for a 16-kbit/s subband coder",
Proc. IEEE ICASSP, pp. 1688-1691, July 1982.

[23] C. Galand and D. Esteban,
"16 kbps sub-band coder incorporating variable overhead information",
Proc. IEEE ICASSP, pp. 1684-1687, July 1982.

[24] P. Noll,
"A comparative study of various quantization schemes for speech encoding",
B.S.T.J., vol. 54, pp. 1597-1614, Nov. 1975.

[25] N.S. Jayant,
"Digital coding of speech waveforms: PCM, DPCM and DM quantizers",
Proc. IEEE, vol. 62, pp. 611-632, May 1974.

[26] J.D. Johnston and R.E. Crochiere,
"An all-digital commentary grade subband coder",
Journal of the audio engineering society, vol. 27, pp. 855-865, Nov. 1979.

[27] E.R. Brigham, M.J. Snaith and D.M. Wilcox,
"Multiplexing for a digital main network",
Post Office Electrical Engineers Journal, vol. 69, pp. 93-102, July 1976.

[28] L.J.D.C. Voorbraak,
"Synchronisatieschakeling voor de subband coder",
Project Report, Telecommunications Division of the Eindhoven University of Technology, the Netherlands, 25 April 1985.

[29] P. Hamel, J. Sourmagne and A. Le Guyader,
"A new dynamic bit allocation scheme for sub-band coding",
IEEE trans. ASSP, vol. 4, pp. 43.3.1-43.3.4, 1985.

[30] T.A. Ramstad,
"Sub-band coder with a simple adaptive bit-allocation algorithm; a possible candidate for digital mobile telephony?",
Proc. IEEE ICASSP, pp. 203-207, July 1982.

[31] C. Galand and D.J. Esteban,
"16 kbps real time QMF sub-band coding implementation",
Proc. IEEE ICASSP, pp. 332-335, April 1980.

[32] R.E. Crochiere,M. Randolph,J.W. Upton and J.D.Johnston,
"Real-time implementation of sub-band coding on a programmable integrated circuit",
Proc. IEEE ICASSP, pp. 455-458, May 1981.

[33] D. Esteban and C. Galand,
"Multiport implementation of real time 16 kbps coder",
Proc. IEEE ICASSP, pp. 459-462, May 1981.

[34] F.S. Yeoh and C.S. Xydeas,
"Split-band coding of speech signals using a transform technique",
Proc. IEEE ICASSP, pp. 1183-1187, Sept. 1984.

[35] R.E. Crochiere and L.R. Rabiner,
"Recent developments in the design and implementation of digital decimators, interpolators and narrow band filters",
IEEE trans. ASSP, vol. 23, pp. 292-295, Oct. 1975.

[36] P. Cummiskey, N.S. Jayant and J.L. Flanagan,
"Adaptive quantization in differential PCM coding of speech",
B.S.T.J., vol. 52, pp. 1105-1118, Sept. 1973.

[37] J. Menez, J.F. Galliano and C. Galand,
"Comparative study of 16 kbps adaptive predictive coders",
Proc. IEEE ICASSP, pp. 619-622, May 1981.

[38] J. Max,
"Quantization for minimum distortion",
IRE Trans. Inf. Theory, vol. IT-6,pp. 7-12, March 1960.

[39] J.D. Gibson, S.K. Jones and J.L. Melsa,
"Sequentially adaptive prediction and coding of speech signals",
IEEE trans. Comm., vol. 22, pp. 1789-1797, Nov. 1974.

[40] T.P. Barnwell,
"Subband coder design incorporating recursive quadrature filters and optimum ADPCM coders",
IEEE trans. ASSP, vol. 30, pp. 751-765, Oct. 1982.

[41] R.W. Schafer and L.R. Rabiner,
"A digital signal processing approach to interpolation",
Proc. IEEE, vol. 61, pp. 692-702, June 1973.

[42] P.C. Millar,
"Mirror filters with minimum delay responses for use in subband coders",
IEEE trans. ASSP, vol. 1, pp. 11.5.1-11.5.4, 1984.

[43] M.B. Donvito and B.W. Schoenherr,
"Subband coding with silence detection",
IEEE trans. ASSP, vol. 4, pp. 37.5.1-37.5.4,1985.

[44] F.K. Soong, R.V. Cox and N.S. Jayant,
"Subband coding of speech using backward adaptive prediction and bit-allocation",
IEEE trans. ASSP, vol. 4, pp. 43.1.1-43.1.4, 1985.

[45] J.H. Derby and C.R. Galand,
"Multirate subband coding applied to digital speech interpolation",
IEEE trans. ASSP, vol. 4, pp. 43.3.1-43.3.4, 1985.

[46] K.K. Paliwal and T. Svendsen,
"A study of three coders (Subband, RELP and MPE) for speech with adaptive white noise",
IEEE trans. ASSP, vol. 4, pp. 43.5.1-43.5.4, 1985.

[47] J.H. Rothweiler,
"Polyphase quadrature filters - A new subband coding technique",
IEEE trans. ASSP, vol. 3, pp. 1280-1283, 1983.

[48] N.S. Jayant,
"Coding speech at low bit rates",
IEEE SPECTRUM, pp. 58-63, Aug. 1986.

[49] C.R. Galand and H.J. Nussbaumer,
"New Quadrature mirror filter structures",
Proc. IEEE ICASSP, vol. 32, pp. 522-530, June 1984.

[50] C.R. Galand and D.J. Esteban,
"Design and evaluation of parallel quadrature mirror filters",
Proc. IEEE ICASSP, pp. 224-227, 1983.

[51] H.J. Nussbaumer,
"Complex quadrature mirror filters",
Proc. IEEE ICASSP, pp. 221-223, 1983.

[52] P.L. Chu,
"Quadrature mirror filter design for an arbitrary number of equal bandwidth channels",
Proc. IEEE ICASSP, vol. 33, pp. 203-218, Febr. 1985.

[53] M.G. Bellanger, G. Bonnerot and M. Coudreuse,
"Digital filtering by polyphase network: application to sample-rate alteration and filter banks",
Proc. IEEE ICASSP, vol. 24, pp. 109-114, April 1986.

[54] P.C. Millar,
"Recursive quadrature mirror filters - criteria specification and design method",
Proc. IEEE ICASSP, vol. 33, pp. 413-420, April 1985.

[55] M.J.T. Smith and T.P. Barnwell,
"Exact reconstruction techniques for tree-structured subband coders",
Proc. IEEE ICASSP, vol. 34, pp. 434-441, June 1986.

[56] E.F. Deprettere and P. Kroon,
"Compression and quantization of speech",
Nederlands Electronica -en Radiogenootschap, deel 49, no. 2, pp. 33-38, 1984.

[57] P.C. Millar,
"A reduced delay 8-band sub-band coder",
Br. Telecom. Technol. J., vol. 3, pp. 57-61, July 1985.

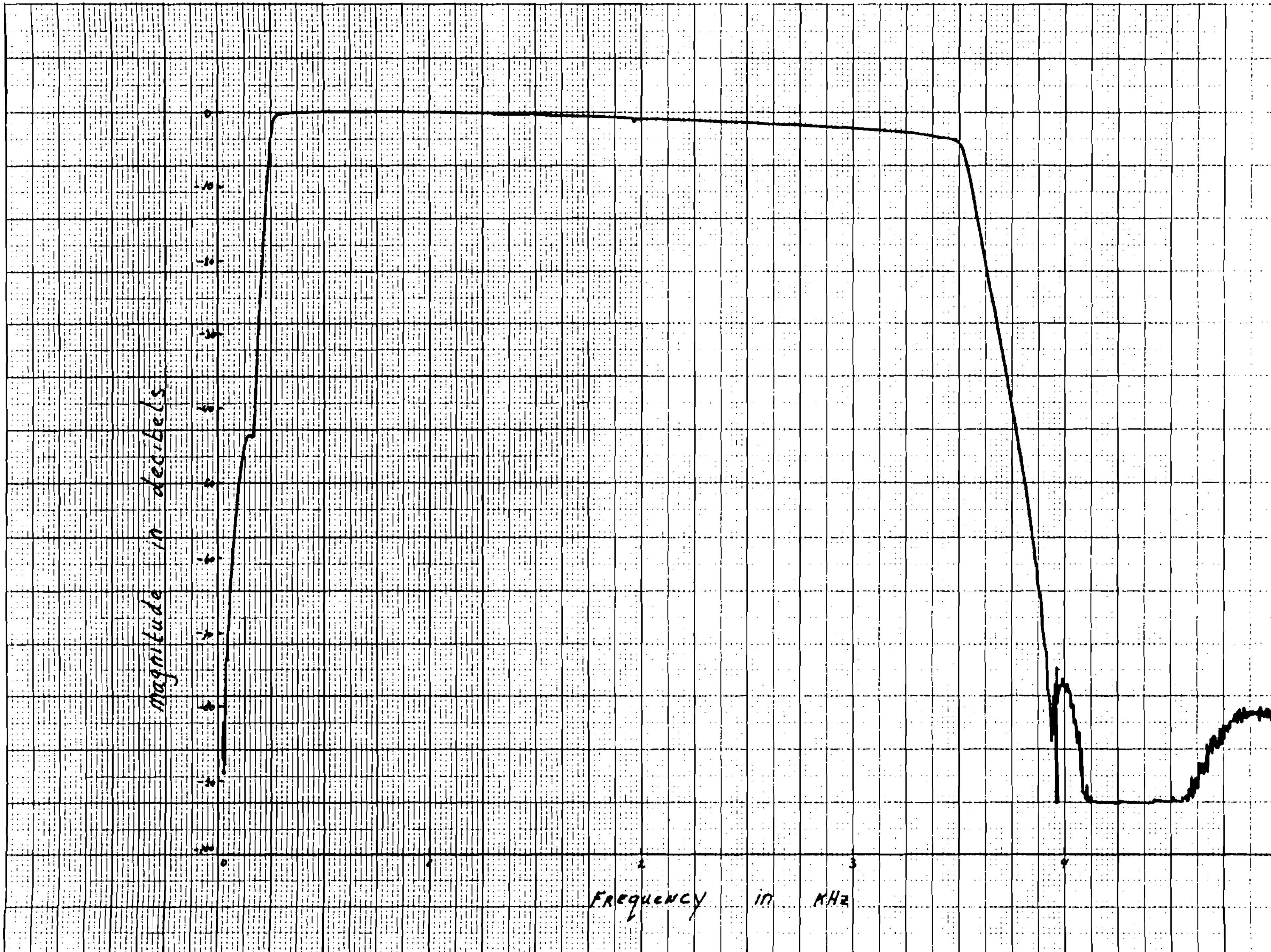
[58] M.R. Sambur,
"Speech algorithm advances promise toll-quality medium-band digitized speech",
Speech Technology, pp. 22-34, Sept. 1982.

[59] J.L. Flanagan and others,
"Speech coding",
IEEE trans. Comm., vol. 27, pp. 710-733, April 1979.

[60] R.E. Crochiere and L.R. Rabiner,
Multirate digital signal processing,
New Jersey: Prentice Hall Inc., Englewood cliffs, 1983.

APPENDIX A

Frequency response of bandlimiting filter


Magnitude in decibels
Frequency in KHz
0
1
2
3
4
5

APPENDIX B

Main program SBC transmitter part

```
****************************************************************************
*                              CODER PROGRAM                               *
*                                                                          *
* THIS MAIN PROGRAM USES THE SUBROUTINES:                                  *
*                                  -QMF FILTERING                          *
*                                  -PCMAQB CODING                          *
*                                  -MULTIPLEXING                           *
* TO REALIZE A 16 kBIT PER SECOND SUBBAND CODER.                           *
* SYSTEM CLOCK RATE: 16 kHz                                                *
*--------------------------------------------------------------------------*
* REPORTS  FOR DETAILS:                                                    *
* PT  REPORT  BY  P.CHITAMU                                                *
* FINAL REPORT BY J.BOLT                                                   *
* FINAL REPORT BY R.KLEUTERS                                               *
*--------------------------------------------------------------------------*
* AUTHOR: ROLAND KLEUTERS                                                  *
* DATE: 29-5-1987                                                          *
****************************************************************************
*
*---------------------------------------------------------------------------
* DATA MEMORY INITIALIZATION
*                                                            default page=0
*---------------------------------------------------------------------------
*
*...........................................................................
*                   GENERAL VARIABLES DATA MEMORY PAGE 1
*...........................................................................
*
CLOCK   EQU  0            CLOCK RATE (16 kHz)
MODE    EQU  1            MODE FOR ANALOG INTERFACE BOARD
AR00    EQU  2            STORAGE PLACE FOR AUXILIARY REGISTER 0
AR01    EQU  3            STORAGE PLACE FOR AUXILIARY REGISTER 1
ACH     EQU  4            STORAGE PLACE FOR HIGH PART OF ACCUMULATOR
ACL     EQU  5            STORAGE PLACE FOR LOW PART OF ACCUMULATOR
TREG    EQU  6            STORAGE PLACE FOR T REGISTER
STATU   EQU  7            STORAGE PLACE FOR STATUS WORD OF TMS32010
*                                              from now on default page
*...........................................................................
*                           GENERAL VARIABLES
*...........................................................................
*
MSTAT   EQU  0            MASK FOR STATE MOD 16
SWITCH  EQU  1            COUNTS FIRST 4 INTERRUPTS; WHEN 0 MUX IS ON    (Q0)
FRAME   EQU  2            COUNTER FOR FRAME LENGTH                       (Q0)
ONE     EQU  3            CHECK BIT                                      (Q0)
STRAT   EQU  4            CONTAINS 0, 1, or 2: VOICED, INTERM. or UNVO.  (Q0)
SSTRAT  EQU  5            BUFFERING                                      (Q0)
STATE   EQU  6            TREE POINTER                                   (Q0)
RES0    EQU  7            INTERMEDIATE CALCULATION RESULTS FOR MAIN ROUTINE
RES1    EQU  8            INTERMEDIATE CALCULATION RESULTS FOR MAIN ROUTINE
PATH    EQU  9            CONTAINS STATE/2=CHANNEL TO PROCESS            (Q0)
*
```

```
*..........................................................................
*                              FIRST QMF STAGE VARIABLES
*..........................................................................
*
*     BUFFER VARIABLES OF LOW1 (0-2 kHz) AND HIGH1 (2-4 kHz)           FORMAT=Q16
A1      EQU  10
B1      EQU  11
C1      EQU  12
*
*     INPUT TO (FILTER PART OF) FIRST QMF STAGE                        FORMAT=Q15
INPF    EQU  13
*
*     DELAY VARIABLES OF LOW1 (0-2 kHz)                                FORMAT=Q15
XF2     EQU  14              AFTER FILTERING INPUT STORED IN XF2, SO
XF3     EQU  15              XF1 NOT NEEDED
XF4     EQU  16
XF5     EQU  17
XF6     EQU  18
XF7     EQU  19
XF8     EQU  20
XF9     EQU  21
XF10    EQU  22
XF11    EQU  23
XF12    EQU  24
XF13    EQU  25
XF14    EQU  26
XF15    EQU  27
XF16    EQU  28
*
*     DELAY VARIABLES OF HIGH1 (2-4 kHz)                               FORMAT=Q15
XF18    EQU  29              AFTER FILTERING INPUT STORED IN XF18, SO
XF19    EQU  30              XF17 NOT NEEDED
XF20    EQU  31
XF21    EQU  32
XF22    EQU  33
XF23    EQU  34
XF24    EQU  35
XF25    EQU  36
XF26    EQU  37
XF27    EQU  38
XF28    EQU  39
XF29    EQU  40
XF30    EQU  41
XF31    EQU  42
XF32    EQU  43
*
*..........................................................................
*                              SECOND QMF STAGE VARIABLES
*..........................................................................
*
*     BUFFER VARIABLES OF LOW2 (0-1 kHz) AND HIGH2 (1-2 kHz)           FORMAT=Q15
A2      EQU  44
B2      EQU  45
C2      EQU  46
*
```

```
*     BUFFER VARIABLES OF LOW3 (3-4 kHz) AND HIGH3 (2-3 kHz)         FORMAT=Q17
A3      EQU  47
B3      EQU  48
C3      EQU  49
*
*     INPUT TO (FILTER PART OF) SECOND QMF STAGE
INPS    EQU  50
*
*     DELAY VARIABLES OF LOW2,HIGH2,LOW3 AND HIGH3 (TIME SHARING)
XS2     EQU  51
XS3     EQU  52
XS4     EQU  53
XS5     EQU  54
XS6     EQU  55
XS7     EQU  56
XS8     EQU  57
*
*..............................................................................
*                              THIRD QMF STAGE VARIABLES
*..............................................................................
*
*     BUFFER VARIABLES OF LOW4 (0-500 Hz) AND HIGH4 (500-1000 Hz)    FORMAT=Q15
A4      EQU  58
B4      EQU  59
C4      EQU  60
*
*     BUFFER VARIABLES OF LOW5 (1500-2000 Hz) AND HIGH5 (1000-1500 Hz) FOR.=Q15
A5      EQU  61
B5      EQU  62
C5      EQU  63
*
*     BUFFER VARIABLES OF LOW7 (2000-2500 Hz) AND HIGH7 (2500-3000 Hz) FOR.=Q18
A7      EQU  64
B7      EQU  65
C7      EQU  66
*
*     INPUT TO (FILTER PART OF) THIRD QMF STAGE
INPT    EQU  67
*
*     DELAY VARIABLES OF LOW4,HIGH4,LOW5,HIGH5,LOW7 AND HIGH7 (TIME SHARING)
XT2     EQU  68
XT3     EQU  69
XT4     EQU  70
XT5     EQU  71
XT6     EQU  72
*..............................................................................
*                            VOICING DECISION VARIABLES
*..............................................................................
*
ENLOW   EQU  73            ENERGY IN 0-2 kHz BAND (Q14)
ENHIGH  EQU  74            ENERGY IN 2-4 kHz BAND (Q16)
*
```

```
*..........................................................................
*                          FIRST QMF STAGE COEFFICIENTS          FORMAT=Q16
*..........................................................................
*
CF0     EQU  75
CF1     EQU  76
CF2     EQU  77
CF3     EQU  78
CF4     EQU  79
CF5     EQU  80
CF6     EQU  81
CF7     EQU  82
CF8     EQU  83
CF9     EQU  84
CF10    EQU  85
CF11    EQU  86
CF12    EQU  87
CF13    EQU  88
CF14    EQU  89
CF15    EQU  90
*
*..........................................................................
*                         SECOND QMF STAGE COEFFICIENTS          FORMAT=Q16
*..........................................................................
*
CS0     EQU  91
CS1     EQU  92
CS2     EQU  93
CS3     EQU  94
CS4     EQU  95
CS5     EQU  96
CS6     EQU  97
CS7     EQU  98
*
*..........................................................................
*                          THIRD QMF STAGE COEFFICIENTS          FORMAT=Q16
*..........................................................................
*
CT0     EQU  99
CT1     EQU  100
CT2     EQU  101
CT3     EQU  102
CT4     EQU  103
CT5     EQU  104
*
*..........................................................................
*                                PCMAQB VARIABLES
*..........................................................................
*
INPCM   EQU  105          INPUT TO PCMAQB CODER
CALC1   EQU  106          GENERAL VARIABLE TO CALCULATE WITH
CALC2   EQU  107          GENERAL VARIABLE TO CALCULATE WITH
NBIT    EQU  108          CONTAINS NR. OF BITS TO QUANTIZE TO            (Q0)
X0      EQU  109          LOWEST POSSIBLE INPUT VALUE OF EXPONENT TABLE (Q14)
X127    EQU  110          HIGHEST POSSIBLE INPUT VALUE OF EXPONENT TABLE(Q14)
STEP    EQU  111          1/(DISTANCE BETWEEN INPUTS OF EXPONENT TABLE)  (Q7)
GAMMA   EQU  112          GAMMA VALUE OF FORMULA (0.99)                 (Q15)
*
```

```
*........................................................................
*                            MULTIPLEXER VARIABLES
*........................................................................
*
COUNT   EQU  113          COUNT FOR BITS TO SEND                        (Q0)
BAND    EQU  114          FLAG, CURRENT SUBBAND PROCESSING              (Q0)
DATA    EQU  115          STORES OUTPUT BITS FOR SIGOUT
SIGOUT  EQU  116          VARIABLE TO WRITE OUT SERIAL DATA FROM INSIDE
DEMO    EQU  117          NEW (FAW or SUBBAND) ACCESS MARK
*
*------------------------------------------------------------------------
* PROGRAM MEMORY INITIALIZATION
*------------------------------------------------------------------------
*
        AORG 0
        B    START        BRANCH TO MAIN ROUTINE
        B    INTRO        BRANCH TO INTERRUPT SERVICE ROUTINE
        AORG 8            OTHERWISE ERRORS WITH TBLW'S
*
*........................................................................
*                            GENERAL CONSTANTS
*........................................................................
*
CCLOCK  DATA  312         CLOCK RATE (16 kHz)
MMODE   DATA  24          MODE FOR ANALOG INTERFACE BOARD
MMSTAT  DATA >000F        MASK FOR STATE MOD 16
*
*........................................................................
*                         SECOND QMF STAGE VARIABLES
*........................................................................
*
*     DELAY VARIABLES OF LOW2 (0-1 kHz)                           FORMAT=Q14
XXS2    DATA 0            AFTER FILTERING INPUT STORED IN XXS2, SO
XXS3    DATA 0            XXS1 NOT NEEDED
XXS4    DATA 0
XXS5    DATA 0
XXS6    DATA 0
XXS7    DATA 0
XXS8    DATA 0
*
*     DELAY VARIABLES OF HIGH2 (1-2 kHz)                          FORMAT=Q14
XXS10   DATA 0            AFTER FILTERING INPUT STORED IN XXS10, SO
XXS11   DATA 0            XXS9 NOT NEEDED
XXS12   DATA 0
XXS13   DATA 0
XXS14   DATA 0
XXS15   DATA 0
XXS16   DATA 0
*
*     DELAY VARIABLES OF LOW3 (3-4 kHz)                           FORMAT=Q16
XXS18   DATA 0            AFTER FILTERING INPUT STORED IN XXS18, SO
XXS19   DATA 0            XXS17 NOT NEEDED
XXS20   DATA 0
XXS21   DATA 0
XXS22   DATA 0
XXS23   DATA 0
XXS24   DATA 0
*
```

```
*     DELAY VARIABLES OF HIGH3 (2-3 kHz)                          FORMAT=Q16
XXS26   DATA 0              AFTER FILTERING INPUT STORED IN XXS26, SO
XXS27   DATA 0              XXS25 NOT NEEDED
XXS28   DATA 0
XXS29   DATA 0
XXS30   DATA 0
XXS31   DATA 0
XXS32   DATA 0
*
*.........................................................................
*                         THIRD QMF STAGE VARIABLES
*.........................................................................
*
*     DELAY VARIABLES OF LOW4 (0-500 Hz)                          FORMAT=Q14
XXT2    DATA 0              AFTER FILTERING INPUT STORED IN XXT2, SO
XXT3    DATA 0              XXT1 NOT NEEEDED
XXT4    DATA 0
XXT5    DATA 0
XXT6    DATA 0
*
*     DELAY VARIABLES OF HIGH4 (500-1000 Hz)                      FORMAT=Q14
XXT8    DATA 0              AFTER FILTERING INPUT STORED IN XXT8, SO
XXT9    DATA 0              XXT7 NOT NEEDED
XXT10   DATA 0
XXT11   DATA 0
XXT12   DATA 0
*
*     DELAY VARIABLES OF LOW5 (1500-2000 Hz)                      FORMAT=Q14
XXT14   DATA 0              AFTER FILTERING INPUT STORED IN XXT14, SO
XXT15   DATA 0              XXT13 NOT NEEDED
XXT16   DATA 0
XXT17   DATA 0
XXT18   DATA 0
*
*     DELAY VARIABLES OF HIGH5 (1000-1500 Hz)                     FORMAT=Q14
XXT20   DATA 0              AFTER FILTERING INPUT STORED IN XXT20, SO
XXT21   DATA 0              XXT19 NOT NEEDED
XXT22   DATA 0
XXT23   DATA 0
XXT24   DATA 0
*
*     DELAY VARIABLES OF LOW7 (2000-2500 Hz)                      FORMAT=Q17
XXT26   DATA 0              AFTER FILTERING INPUT STORED IN XXT26, SO
XXT27   DATA 0              XXT25 NOT NEEDED
XXT28   DATA 0
XXT29   DATA 0
XXT30   DATA 0
*
*     DELAY VARIABLES OF HIGH7 (2500-3000 Hz)                     FORMAT=Q17
XXT32   DATA 0              AFTER FILTERING INPUT STORED IN XXT32, SO
XXT33   DATA 0              XXT31 NOT NEEDED
XXT34   DATA 0
XXT35   DATA 0
XXT36   DATA 0
*
```

```
*.............................................................................
*                        FIRST QMF STAGE COEFFICIENTS              FORMAT=Q16
*.............................................................................
*
CCF0    DATA  45
CCF1    DATA -92
CCF2    DATA -83
CCF3    DATA  277
CCF4    DATA  93
CCF5    DATA -620
CCF6    DATA -9
CCF7    DATA  1178
CCF8    DATA -274
CCF9    DATA -2047
CCF10   DATA  955
CCF11   DATA  3470
CCF12   DATA -2579
CCF13   DATA -6541
CCF14   DATA  8425
CCF15   DATA  30566
*
*.............................................................................
*                        SECOND QMF STAGE COEFFICIENTS             FORMAT=Q16
*.............................................................................
*
CCS0    DATA  69
CCS1    DATA -331
CCS2    DATA -170
CCS3    DATA  1812
CCS4    DATA -633
CCS5    DATA -5924
CCS6    DATA  6409
CCS7    DATA  31525
*
*.............................................................................
*                         THIRD QMF STAGE COEFFICIENTS             FORMAT=Q16
*.............................................................................
*
CCT0    DATA -250
CCT1    DATA  1236
CCT2    DATA -178
CCT3    DATA -5551
CCT4    DATA  5798
CCT5    DATA  31745
*
*.............................................................................
*                             PCMAQB's  TABLES
*.............................................................................
*
*     (1-GAMMA)*DC FOR 6 SUBBANDS                                  FORMAT=Q22
OGD1    DATA -10486         SUBBAND 1
DUM1    DATA  0
OGD4    DATA -13642         SUBBAND 4
OGD5    DATA -23112         SUBBAND 5
OGD2    DATA -10486         SUBBAND 2
DUM2    DATA  0
OGD3    DATA -13642         SUBBAND 3
OGD6    DATA -23112         SUBBAND 6
```

```
*
*     D(N) FOR 6 SUBBANDS                                          FORMAT=Q14
DN1     DATA -13563         SUBBAND 1
DUM3    DATA 0
DN4     DATA -14796         SUBBAND 4
DN5     DATA -18495         SUBBAND 5
DN2     DATA -13563         SUBBAND 2
DUM4    DATA  0
DN3     DATA -14796         SUBBAND 3
DN6     DATA -18495         SUBBAND 6
*
*     BIT ALLOCATION FOR VOICED STRATEGY (0)                       FORMAT=Q0
BAVOI1  DATA 4              SUBBAND 1
DUM5    DATA 0
BAVOI4  DATA 2              SUBBAND 4
BAVOI5  DATA 2              SUBBAND 5
BAVOI2  DATA 4              SUBBAND 2
DUM6    DATA 0
BAVOI3  DATA 3              SUBBAND 3
BAVOI6  DATA 0              SUBBAND 6
*
*     BIT ALLOCATION FOR INTERMEDIATE STRATEGY (1)                 FORMAT=Q0
BAINT1  DATA 4              SUBBAND 1
DUM7    DATA 0
BAINT4  DATA 2              SUBBAND 4
BAINT5  DATA 2              SUBBAND 5
BAINT2  DATA 3              SUBBAND 2
DUM8    DATA 0
BAINT3  DATA 2              SUBBAND 3
BAINT6  DATA 2              SUBBAND 6
*
*     BIT ALLOCATION FOR UNVOICED STRATEGY (2)                     FORMAT=Q0
BAUNV1  DATA 2              SUBBAND 1
DUM9    DATA 0
BAUNV4  DATA 3              SUBBAND 4
BAUNV5  DATA 2              SUBBAND 5
BAUNV2  DATA 3              SUBBAND 2
DUM10   DATA 0
BAUNV3  DATA 3              SUBBAND 3
BAUNV6  DATA 2              SUBBAND 6
*
*     STORAGE OF CODED SUBBAND SAMPLES                             FORMAT=Q0
COD1    DATA 0              SUBBAND 1
DUM11   DATA 0
COD4    DATA 0              SUBBAND 4
COD5    DATA 0              SUBBAND 5
COD2    DATA 0              SUBBAND 2
DUM12   DATA 0
COD3    DATA 0              SUBBAND 3
COD6    DATA 0              SUBBAND 6
*
*     SATURATION LEVELS FOR QUANTIZER                              FORMAT=Q0
SAT1    DATA 1              2 BITS
SAT2    DATA 3              3 BITS
SAT3    DATA 7              4 BITS
*
```

```
*     QUANTIZER CONSTANTS
SSTEP   DATA  19637         1/(DISTANCE BETWEEN INPUTS OF EXPONENT TABLE) (Q7)
GGAMMA  DATA  32440         GAMMA=0.99                                    (Q15)
*
*     LOGTABLE 2 BIT CASE                                          FORMAT=Q18
LOG20   DATA  18268
LOG21   DATA -4626
LOG22   DATA -4626
LOG23   DATA  18268
DUM13   DATA  0
DUM14   DATA  0
DUM15   DATA  0
DUM16   DATA  0
DUM17   DATA  0
DUM18   DATA  0
DUM19   DATA  0
DUM20   DATA  0
DUM21   DATA  0
DUM22   DATA  0
*
*     LOGTABLE 3 BIT CASE                                          FORMAT=Q18
LOG30   DATA  11540
LOG31   DATA  0
LOG32   DATA  0
LOG33   DATA -4626
LOG34   DATA -4626
LOG35   DATA  0
LOG36   DATA  0
LOG37   DATA  11540
DUM23   DATA  0
DUM24   DATA  0
DUM25   DATA  0
DUM26   DATA  0
*
*     LOGTABLE 4 BIT CASE                                          FORMAT=Q18
LOG40   DATA  24918
LOG41   DATA  19728
LOG42   DATA  13377
LOG43   DATA  5189
LOG44   DATA -2999
LOG45   DATA -2999
LOG46   DATA -2999
LOG47   DATA -2999
LOG48   DATA -2999
LOG49   DATA -2999
LOG410  DATA -2999
LOG411  DATA -2999
LOG412  DATA  5189
LOG413  DATA  13377
LOG414  DATA  19728
LOG415  DATA  24918
*
```

```
*     LOWEST AND HIGHEST POSSIBLE INPUTS OF EXPONENT TABLE          FORMAT=Q14
XX10    DATA -13563         SUBBAND 1
XX1127  DATA  0             SUBBAND 1
DUM27   DATA  0
DUM28   DATA  0
XX40    DATA -14796         SUBBAND 4
XX4127  DATA -1233          SUBBAND 4
XX50    DATA -18495         SUBBAND 5
XX5127  DATA -4932          SUBBAND 5
XX20    DATA -13563         SUBBAND 2
XX2127  DATA  0             SUBBAND 2
DUM29   DATA  0
DUM30   DATA  0
XX30    DATA -14796         SUBBAND 3
XX3127  DATA -1233          SUBBAND 3
XX60    DATA -18495         SUBBAND 6
XX6127  DATA -4932          SUBBAND 6
*
*     EXPONENT TABLE OF CODER
*     FORMATS ARE: SUBBAND 1  Q4
*                  SUBBAND 2  Q4
*                  SUBBAND 3  Q3
*                  SUBBAND 4  Q3
*                  SUBBAND 5  Q0
*                  SUBBAND 6  Q0
*
CEXPO   DATA 32767
        DATA 30859
        DATA 29060
        DATA 27367
        DATA 25772
        DATA 24271
        DATA 22856
        DATA 21525
        DATA 20270
        DATA 19089
        DATA 17977
        DATA 16929
        DATA 15943
        DATA 15014
        DATA 14139
        DATA 13315
        DATA 12539
        DATA 11809
        DATA 11121
        DATA 10473
        DATA 9862
        DATA 9288
        DATA 8746
        DATA 8237
        DATA 7757
        DATA 7305
        DATA 6879
        DATA 6478
        DATA 6101
        DATA 5745
        DATA 5411
        DATA 5095
```

```
DATA 4798
DATA 4519
DATA 4256
DATA 4008
DATA 3774
DATA 3554
DATA 3347
DATA 3152
DATA 2968
DATA 2795
DATA 2632
DATA 2479
DATA 2335
DATA 2199
DATA 2070
DATA 1950
DATA 1836
DATA 1729
DATA 1628
DATA 1534
DATA 1444
DATA 1360
DATA 1281
DATA 1206
DATA 1136
DATA 1070
DATA 1007
DATA 949
DATA 893
DATA 841
DATA 792
DATA 746
DATA 703
DATA 662
DATA 623
DATA 587
DATA 553
DATA 520
DATA 490
DATA 462
DATA 435
DATA 409
DATA 385
DATA 363
DATA 342
DATA 322
DATA 303
DATA 286
DATA 269
DATA 253
DATA 238
DATA 225
DATA 211
DATA 199
DATA 188
DATA 177
DATA 166
DATA 157
```

```
        DATA 148
        DATA 139
        DATA 131
        DATA 123
        DATA 116
        DATA 109
        DATA 103
        DATA 97
        DATA 91
        DATA 86
        DATA 81
        DATA 76
        DATA 72
        DATA 68
        DATA 64
        DATA 60
        DATA 56
        DATA 53
        DATA 50
        DATA 47
        DATA 44
        DATA 42
        DATA 39
        DATA 37
        DATA 35
        DATA 33
        DATA 31
        DATA 29
        DATA 27
        DATA 26
        DATA 24
        DATA 23
        DATA 22
        DATA 20
        DATA 19
        DATA 18
        DATA 17
        DATA 16
*
*-----------------------------------------------------------------------------
* MAIN ROUTINE
*-----------------------------------------------------------------------------
*
*.............................................................................
*                               INITIALIZATIONS
*.............................................................................
*
START   DINT
*
*     INITIALIZE STATUS BITS
        LARP 0
        LDPK 0
        SOVM
*
```

```
*       INITIALIZE GENERAL VARIABLES OF DATA MEMORY PAGE 1
         LDPK 1
         LACK CCLOCK
         TBLR CLOCK          CLOCK RATE (16 kHz)
         LACK MMODE
         TBLR MODE           MODE FOR ANALOG INTERFACE BOARD
         LDPK 0
*
*       INITIALIZE GENERAL VARIABLES OF DATA MEMORY PAGE 0
         LACK MMSTAT
         TBLR MSTAT          MASK FOR STATE MOD 16
         LACK 5              NO. OF WAIT CYCLI BEFOR MULTIPLEXER START
         SACL SWITCH         COUNTER FOR MULTIPLEXER ON or OFF, NOW OFF
         ZAC
         SACL FRAME          STATUS FOR CODER
         LACK 1
         SACL ONE            '1' FOR LOGICAL MANIPULATIONS
         SACL STRAT          BY DEFAULT LOAD INTERMEDIATE STRATEGY
         SACL SSTRAT         ALSO INTERMEDIATE STRATEGY FOR BUFFER
         ZAC
         SACL STATE          INIT TREE POINTER
*
*       INITIALIZE PCMAQB VARIABLES
         LACK SSTEP
         TBLR STEP           1/(DISTANCE BETWEEN INPUTS OF EXPONENT TABLE)
         LACK GGAMMA
         TBLR GAMMA          GAMMA VALUE OF FORMULA (0.99)
*
*       INITIALIZE MULTIPLEXER VARIABLES
         ZAC
         SACL BAND           STATUS FOR MULTIPLEXER
         LACK >FF
         SACL DEMO           NEW ACCESS MARK
*
*       LOAD QMF COEFFICIENTS INTO DATA RAM
         LARK 1,CT5
         LARK 0,29
         LACK CCT5
LOAD     LARP 1
         TBLR *-,0
         SUB  ONE
         BANZ LOAD
*
*       INITIALIZE ANALOG INTERFACE BOARD (AIB)
         LDPK 1
         OUT  MODE,0         PROGRAM AIB
         OUT  CLOCK,1        SET CLOCK RATE OF AIB
         LDPK 0
         OUT  RESO,2         CLEAR AD1S (PROTECTION AGAINST RESET INTERRUPTS)
         OUT  RESO,3         CLEAR AD2S (PROTECTION AGAINST RESET INTERRUPTS)
*
*       INITIALIZATION AND VARIABLE LOADING DONE, NOW BEGIN
         EINT
*
```

```
*    DISCARD FIRST TWO INTERRUPTS TO GET A DEFINED STARTING POINT
ACKNO   ZALS STATE
        BZ   ACKNO
        LACK 1
        XOR  STATE
        BZ   ACKNO
        ZAC
        SACL STATE          INIT TREE POINTER
*
*.........................................................................
*                                   AORTA
*.........................................................................
*
*    ODD CYCLE OF 16 kHz CLOCK MUST JUST HAVE PASSED
WAITEV  LAC  STATE
        AND  ONE
        BNZ  WAITEV
*    EVEN CYCLE OF 16 kHz CLOCK HAS PASSED (BIT 0 OF STATE=0)
WAIT    LAC  STATE
        AND  ONE
        BZ   WAIT
*    ODD CYCLE OF 16 kHz CLOCK HAS JUST PASSED (BIT 0 OF STATE=1)
*
*    INPUT SAMPLE RATE REDUCED FROM 16 kHz (2*8 kHz) TO 8 kHz BY OMITTING
*    EVERY EVEN INPUT SAMPLE
*
*    DURING THE PROCESSING OF A SUBBAND SAMPLE STATE CAN CHANGE AS A
*    RESULT OF AN INTERRUPT, SO DEFINE PATH=STATE/2
        LAC  STATE,15
        SACH PATH
*
*    ACTUAL SUBBAND CODING
        CALL QMF
        CALL PCMAQB
        B    WAITEV
*
*-------------------------------------------------------------------------
* INTERRUPT SERVICE ROUTINE
*-------------------------------------------------------------------------
*
INTRO   DINT
*
*    SAVE STATUS OF TMS32010
        SST  STATU
        MPY  ONE            T REGISTER TO P REGISTER
        LDPK 1
        SAR  0,AR00
        SAR  1,AR01
        SACH ACH
        SACL ACL
        PAC
        SACL TREG
        LDPK 0
*
```

```
*     STATE  UPDATE
        LAC  STATE
        ADD  ONE
        AND  MSTAT          STATE MOD 16
        SACL STATE
*
*     SAMPLE INPUT (16 kHz = 2*8 kHz)
        IN   INPF,2         ONE OF TWO SAMPLES IS USED LATER
*
*     CHECK IF MULTIPLEXER IS ON (SWITCH=0); IF NOT DECREMENT SWITCH
*     AND RETURN TO PLACE OF INTERRUPT
        ZALS SWITCH
        BZ   MUXER
        SUB  ONE
        SACL SWITCH
        B    RSSTAT
*
*     RUN THROUGH MULTIPLEXER
MUXER   CALL MUX
*
*     RESTORE STATUS OF TMS32010
RSSTAT  LDPK 1
        LAR  0,AR00
        LAR  1,AR01
        ZALH ACH
        ADDS ACL
        LT   TREG
        LST  STATU
*
*     RETURN TO PLACE OF INTERRUPT AND CONTINUE
        EINT
        RET
*
*                                CODER PROGRAM DONE
*=============================================================================
```

APPENDIX C

Subroutine QMF

```
******************************************************************************
*                              QMF FILTERING                                 *
*                                                                            *
* THIS SUBROUTINE INCORPORATES:                                              *
*                            -QMF FILTER TREE                                *
*                            -ENERGY MEASUREMENT                             *
*                            -VOICING STRATEGY DECISION                      *
*                            -TAKE OVER OF NEW VOICING STRATEGY              *
*                             FOR THE PCM CODING                             *
*                                                                            *
* INPUT:  8 kHz SAMPLES (1 STREAM)                                           *
* OUTPUT: FAW + 1 kHz SAMPLES (8 STREAMS)                                    *
*----------------------------------------------------------------------------*
* SUBBAND PROCESSING : 1,8,4,5,2,7,3,6                                       *
*                                                                            *
* BIT ALLOCATION :                                                           *
*                 ______________1___2___3___4___5___6___7___8_____           *
*                 VOICED        4   4   3   2   2   0   0   0                *
*                 INTERMEDIATE  4   3   2   2   2   2   0   0                *
*                 UNVOICED      2   3   3   3   2   2   0   0                *
* FAW's :                                                                    *
*        VOICED >76        INTERMEDIATE >96         UNVOICED >CC             *
*----------------------------------------------------------------------------*
* AUTHOR: ROLAND KLEUTERS                                                    *
* DATE: 29-5-1987                                                            *
******************************************************************************
*
*-----------------------------------------------------------------------------
* MACRO'S
*-----------------------------------------------------------------------------
*
*    MACRO TO REALIZE FILTERING IN LOW BRANCH OF FIRST QMF STAGE
FLTFLO  $MACRO X
        ZAC
        LT   XF16
        MPY  CF0
        LTD  XF15
        MPY  CF2
        LTD  XF14
        MPY  CF4
        LTD  XF13
        MPY  CF6
        LTD  XF12
        MPY  CF8
        LTD  XF11
        MPY  CF10
        LTD  XF10
        MPY  CF12
        LTD  XF9
        MPY  CF14
        LTD  XF8
        MPY  CF15
        LTD  XF7
        MPY  CF13
        LTD  XF6
        MPY  CF11
        LTD  XF5
        MPY  CF9
```

```
        LTD  XF4
        MPY  CF7
        LTD  XF3
        MPY  CF5
        LTD  XF2
        MPY  CF3
        LTA  INPF           8 kHz INPUT TO FIRST QMF STAGE
        MPY  CF1
        APAC
        SACH :X.S:,1
        LAC  INPF
        SACL XF2
        $END
*
*     MACRO TO REALIZE FILTERING IN HIGH BRANCH OF FIRST QMF STAGE
FLTFHI  $MACRO X
        ZAC
        LT   XF32
        MPY  CF1
        LTD  XF31
        MPY  CF3
        LTD  XF30
        MPY  CF5
        LTD  XF29
        MPY  CF7
        LTD  XF28
        MPY  CF9
        LTD  XF27
        MPY  CF11
        LTD  XF26
        MPY  CF13
        LTD  XF25
        MPY  CF15
        LTD  XF24
        MPY  CF14
        LTD  XF23
        MPY  CF12
        LTD  XF22
        MPY  CF10
        LTD  XF21
        MPY  CF8
        LTD  XF20
        MPY  CF6
        LTD  XF19
        MPY  CF4
        LTD  XF18
        MPY  CF2
        LTA  INPF           8 kHz INPUT TO FIRST QMF STAGE
        MPY  CF0
        APAC
        SACH :X.S:,1
        LAC  INPF
        SACL XF18
        $END
*
```

```
*     MACRO TO REALIZE FILTERING IN LOW BRANCH OF SECOND QMF STAGE
FLTSLO  $MACRO X
        ZAC
        LT   XS8
        MPY  CS0
        LTD  XS7
        MPY  CS2
        LTD  XS6
        MPY  CS4
        LTD  XS5
        MPY  CS6
        LTD  XS4
        MPY  CS7
        LTD  XS3
        MPY  CS5
        LTD  XS2
        MPY  CS3
        LTD  INPS           4 kHz INPUT TO SECOND QMF STAGE
        MPY  CS1
        APAC
        SACH :X.S:,1
        $END
*
*     MACRO TO REALIZE FILTERING IN HIGH BRANCH OF SECOND QMF STAGE
FLTSHI  $MACRO X
        ZAC
        LT   XS8
        MPY  CS1
        LTD  XS7
        MPY  CS3
        LTD  XS6
        MPY  CS5
        LTD  XS5
        MPY  CS7
        LTD  XS4
        MPY  CS6
        LTD  XS3
        MPY  CS4
        LTD  XS2
        MPY  CS2
        LTD  INPS           4 kHz INPUT TO SECOND QMF STAGE
        MPY  CS0
        APAC
        SACH :X.S:,1
        $END
*
*     MACRO TO REALIZE FILTERING IN LOW BRANCH OF THIRD QMF STAGE
FLTTLO  $MACRO X
        ZAC
        LT   XT6
        MPY  CT0
        LTD  XT5
        MPY  CT2
        LTD  XT4
        MPY  CT4
        LTD  XT3
        MPY  CT5
        LTD  XT2
```

```
        MPY  CT3
        LTD  INPT           2 kHz INPUT TO THIRD QMF STAGE
        MPY  CT1
        APAC
        SACH :X.S:,1
        $END
*
*    MACRO TO REALIZE FILTERING IN HIGH BRANCH OF THIRD QMF STAGE
FLTTHI  $MACRO X
        ZAC
        LT   XT6
        MPY  CT1
        LTD  XT5
        MPY  CT3
        LTD  XT4
        MPY  CT5
        LTD  XT3
        MPY  CT4
        LTD  XT2
        MPY  CT2
        LTD  INPT           2 kHz INPUT TO THIRD QMF STAGE
        MPY  CT0
        APAC
        SACH :X.S:,1
        $END
*
*    MACRO TO LOAD DELAY VARIABLES OF SECOND QMF STAGE
LOADDS  $MACRO X
        LACK :X.S:
        LARK 0,XS2
        TBLR *+
        ADD  ONE
        TBLR *+
        ADD  ONE
        TBLR *+
        ADD  ONE
        TBLR *+
        ADD  ONE
        TBLR *+
        ADD  ONE
        TBLR *+
        ADD  ONE
        TBLR *
        $END
*
*    MACRO TO SAVE DELAY VARIABLES OF SECOND QMF STAGE
SAVEDS  $MACRO X
        LACK :X.S:
        LARK 0,XS2
        TBLW *+
        ADD  ONE
        TBLW *+
        ADD  ONE
        TBLW *+
        ADD  ONE
        TBLW *+
        ADD  ONE
        TBLW *+
```

```
        ADD  ONE
        TBLW *+
        ADD  ONE
        TBLW *
        $END
*
*    MACRO TO LOAD DELAY VARIABLES OF THIRD QMF STAGE
LOADDT  $MACRO X
        LACK :X.S:
        LARK 0,XT2
        TBLR *+
        ADD  ONE
        TBLR *+
        ADD  ONE
        TBLR *+
        ADD  ONE
        TBLR *+
        ADD  ONE
        TBLR *
        $END
*
*    MACRO TO SAVE DELAY VARIABLES OF THIRD QMF STAGE
SAVEDT  $MACRO X
        LACK :X.S:
        LARK 0,XT2
        TBLW *+
        ADD  ONE
        TBLW *+
        ADD  ONE
        TBLW *+
        ADD  ONE
        TBLW *+
        ADD  ONE
        TBLW *
        $END
*
*---------------------------------------------------------------------------
* MACRO DEFINITION DONE; MAIN PART OF SUBROUTINE BEGINS
*---------------------------------------------------------------------------
*
QMF     NOP                 ENTRY POINT
*
*    SELECT LOW1 OR HIGH1 FOR FIRST QMF STAGE
        LAC  ONE            BIT 0=0 or 1
        AND  PATH
        BZ   LOW1
        B    HIGH1
*
*---------------------------------------------------------------------------
* FIRST QMF STAGE:
*                 LOW1  0kHz-2kHz
*                 HIGH1 2kHz-4kHz
*---------------------------------------------------------------------------
*
LOW1    DMOV A1             DOUBLE BUFFER
*
```

```
*       OUTPUT CALCULATION
          LAC  C1,14
          ADD  B1,14
          SACH INPS             4 kHz INPUT TO SECOND QMF STAGE
*
*       FILTER 0-2 kHz
          FLTFLO A1
*
*       ENERGY SUMMATION, SUM X(n) SQUARED
          LT   INPS
          MPY  INPS
          PAC                   ACCU CONTAINS INPS*INPS IN Q28 FORMAT
          ADD  ENLOW,14         ACCU CONTAINS ENLOW IN Q28 FORMAT
          SACL RES0             MULTIPLY ACCU WITH 4
          SACH RES1
          ADD  RES0
          ADDH RES1
          ADD  RES0
          ADDH RES1
          ADD  RES0
          ADDH RES1             ACCU CONTAINS ENLOW IN Q30 FORMAT
          SACH ENLOW            ENLOW IN Q14 FORMAT
*
*       SELECT LOW2 OR HIGH2 FOR SECOND QMF STAGE
          LAC  ONE,1
          AND  PATH
          BZ   LOW2
          B    HIGH2
*
HIGH1     NOP                   NO DOUBLE BUFFER
*
*       OUTPUT CALCULATION
          ZALH C1
          SUBH B1
          SACH INPS             4 kHz INPUT TO SECOND QMF STAGE
*
*       FILTER 2-4 kHz
          FLTFHI C1
*
*       ENERGY SUMMATION, SUM X(n) SQUARED
          LT   INPS
          MPY  INPS
          PAC                   ACCU CONTAINS INPS*INPS IN Q32 FORMAT
          ADDH ENHIGH           ACCU CONTAINS ENHIGH IN Q32 FORMAT
          SACH ENHIGH           ENHIGH IN Q16 FORMAT
*
*       SELECT LOW3 OR HIGH3 FOR SECOND QMF STAGE
          LAC  ONE,1
          AND  PATH
          BZ   LOW3
          B    HIGH3
*
```

```
*-----------------------------------------------------------------------
* SECOND QMF STAGE:
*                 LOW2  0kHz-1kHz
*                 HIGH2 1kHz-2kHz
*                 LOW3  3kHz-4kHz
*                 HIGH3 2kHz-3kHz
*-----------------------------------------------------------------------
*
LOW2    DMOV A2             DOUBLE BUFFER
*
*     OUTPUT CALCULATION
        LAC  C2,15
        ADD  B2,15
        SACH INPT           2 kHz INPUT TO THIRD QMF STAGE
*
*     LOAD DELAY VARIABLES OF LOW2
        LOADDS XXS2
*
*     FILTER 0-1 kHz
        FLTSLO A2
*
*     SAVE DELAY VARIABLES OF LOW2
        SAVEDS XXS2
*
*     SELECT LOW4 OR HIGH4 FOR THIRD QMF STAGE
        LAC  ONE,2
        AND  PATH
        BZ   LOW4
        B    HIGH4
*
HIGH2   NOP                 NO DOUBLE BUFFER
*
*     OUTPUT CALCULATION
        LAC  C2,15
        SUB  B2,15
        SACH INPT           2 kHz INPUT TO THIRD QMF STAGE
*
*     LOAD DELAY VARIABLES OF HIGH2
        LOADDS XXS10
*
*     FILTER 1-2 kHz
        FLTSHI C2
*
*     SAVE DELAY VARIABLES OF HIGH2
        SAVEDS XXS10
*
*     SELECT LOW5 OR HIGH5 FOR THIRD QMF STAGE
        LAC  ONE,2
        AND  PATH
        BZ   LOW5
        B    HIGH5
*
LOW3    DMOV A3             DOUBLE BUFFER
*
*     OUTPUT CALCULATION
        ZALH C3
        ADDH B3
        SACH INPT           2 kHz INPUT TO THIRD QMF STAGE
```

```
*
*     LOAD DELAY VARIABLES OF LOW3
       LOADDS XXS18
*
*     FILTER 3-4 kHz
       FLTSLO A3
*
*     SAVE DELAY VARIABLES OF LOW3
       SAVEDS XXS18
*
*     THE 3-4 kHz SUBBAND IS NOT CODED, SO FURTHER SPLITTING IS NOT NECESSARY
*
*     VOICING DECISION?
*     FRAME HAS TO BE 3 AND PATH HAS TO BE 5; INTERRUPT 75
       LACK 3
       XOR  FRAME
       BNZ  NOVODE         FRAME=3?
       LACK 5
       XOR  PATH           FRAME=5?
       BZ   VOIDEC
NOVODE RET                 NOT INTERRUPT 75; NO VOICING DECISION; QMF DONE
*
HIGH3  NOP                 NO DOUBLE BUFFER
*
*     OUTPUT CALCULATION
       ZALH C3
       SUBH B3
       SACH INPT           2 kHz INPUT TO THIRD QMF STAGE
*
*     LOAD DELAY VARIABLES OF HIGH3
       LOADDS XXS26
*
*     FILTER 2-3 kHz
       FLTSHI C3
*
*     SAVE DELAY VARIABLES OF HIGH3
       SAVEDS XXS26
*
*     SELECT LOW7 OR HIGH7 FOR THIRD QMF STAGE
       LAC  ONE,2
       AND  PATH
       BZ   LOW7
       B    HIGH7
*
*----------------------------------------------------------------------------
* THIRD QMF STAGE:
*              LOW4  0-500     Hz
*              HIGH4 500-1000  Hz
*              LOW5  1500-2000 Hz
*              HIGH5 1000-1500 Hz
*              LOW7  2000-2500 Hz
*              HIGH7 2500-3000 Hz
*              MAKE A VOICING DECISION
*              TAKE OVER NEW VOICING STRATEGY
*----------------------------------------------------------------------------
*
LOW4   DMOV A4             DOUBLE BUFFER
*
```

```
*     OUTPUT CALCULATION
        LAC  C4,15
        ADD  B4,15
        SACH INPCM          1 kHz INPUT TO PCM CODER
*
*     LOAD DELAY VARIABLES OF LOW4
        LOADDT XXT2
*
*     FILTER 0-500 Hz
        FLTTLO A4
*
*     SAVE DELAY VARIABLES OF LOW4
        SAVEDT XXT2
*
*     QMF DONE;
*     BEFORE EXIT, UPDATE FRAME AND TAKE OVER NEW STRATEGY IF END OF FRAME
        LAR  0,FRAME
        BANZ SFRAME
        LACK 7              END OF FRAME; PREPARE NEXT FRAME
        SACL FRAME
        DMOV STRAT          TAKE OVER NEW VOICING STRATEGY
        RET
SFRAME  SAR  0,FRAME        NOT END OF FRAME; PREPARE NEXT DATABLOCK
        RET
*
HIGH4   NOP                 NO DOUBLE BUFFER
*
*     OUTPUT CALCULATION
        LAC  C4,15
        SUB  B4,15
        SACH INPCM          1 kHz INPUT TO PCM CODER
*
*     LOAD DELAY VARIABLES OF HIGH4
        LOADDT XXT8
*
*     FILTER 500-1000 Hz
        FLTTHI C4
*
*     SAVE DELAY VARIABLES OF HIGH4
        SAVEDT XXT8
*
*     QMF DONE
        RET
*
LOW5    DMOV A5             DOUBLE BUFFER
*
*     OUTPUT CALCULATION
        ZALH C5
        ADDH B5
        SACH INPCM          1 kHz INPUT TO PCM CODER
*
*     LOAD DELAY VARIABLES OF LOW5
        LOADDT XXT14
*
*     FILTER 1500-2000 Hz
        FLTTLO A5
*
```

```
*       SAVE DELAY VARIABLES OF LOW5
          SAVEDT XXT14
*
*       QMF DONE
          RET
*
HIGH5     NOP                  NO DOUBLE BUFFER
*
*       OUTPUT CALCULATION
          ZALH C5
          SUBH B5
          SACH INPCM           1 kHz INPUT TO PCM CODER
*
*       LOAD DELAY VARIABLES OF HIGH5
          LOADDT XXT20
*
*       FILTER 1000-1500 Hz
          FLTTHI C5
*
*       SAVE DELAY VARIABLES OF HIGH5
          SAVEDT XXT20
*
*       QMF DONE
          RET
*
LOW7      DMOV A7              DOUBLE BUFFER
*
*       OUTPUT CALCULATION
          ZALH C7
          ADDH B7
          SACH INPCM           1 kHz INPUT TO PCM CODER
*
*       LOAD DELAY VARIABLES OF LOW7
          LOADDT XXT26
*
*       FILTER 2000-2500 Hz
          FLTTLO A7
*
*       SAVE DELAY VARIABLES OF LOW7
          SAVEDT XXT26
*
*       QMF DONE
          RET
*
HIGH7     NOP                  NO DOUBLE BUFFER
*
*       OUTPUT CALCULATION
          ZALH C7
          SUBH B7
          SACH INPCM           1 kHz INPUT TO PCM CODER
*
*       LOAD DELAY VARIABLES OF HIGH7
          LOADDT XXT32
*
*       FILTER 2500-3000 Hz
          FLTTHI C7
*
```

```
*     SAVE DELAY VARIABLES OF HIGH7
        SAVEDT XXT32
*
*     QMF DONE
        RET
*
*     MAKE A VOICING DECISION
VOIDEC  NOP
*
*     IF 20ENHIGH-ENLOW LESS THAN OR EQUAL TO ZERO THEN VOICED
        LT   ENHIGH
        MPYK +20
        PAC
        SUB  ENLOW,2
        BLEZ SETVOI
*
*     IF 3ENHIGH-2ENLOW GREATER THAN OR EQUAL TO ZERO THEN UNVOICED
        LT   ENHIGH
        MPYK +3
        PAC
        SUB  ENLOW,2
        SUB  ENLOW,2
        BGEZ SETUN
*
*     IF 10ENHIGH-ENLOW LESS THAN OR EQUAL TO ZERO AND PREVIOUS VOICING
*     WAS VOICED THEN VOICED
        LAC  STRAT
        BNZ  PREUNV
        LT   ENHIGH
        MPYK +10
        PAC
        SUB  ENLOW,2
        BLEZ SETVOI
*
*     IF 3ENHIGH-ENLOW GREATER THAN OR EQUAL TO ZERO AND PREVIOUS VOICING
*     WAS UNVOICED THEN UNVOICED
PREUNV  LACK 2
        XOR  STRAT
        BNZ  SETINT
        LT   ENHIGH
        MPYK 3
        PAC
        SUB  ENLOW,2
        BGEZ SETUN
*
*     ELSE NEW STRATEGY IS INTERMEDIATE STRATEGY
*
*     SET INTERMEDIATE STRATEGY AND CLEAR VOICING DECISION BUFFERS
SETINT  LACK 1
        SACL STRAT
        ZAC
        SACL ENLOW
        SACL ENHIGH
        RET                 QMF DONE
*
```

```
*     SET VOICED STRATEGY AND CLEAR VOICING DECISION BUFFERS
SETVOI  ZAC
        SACL STRAT
        SACL ENLOW
        SACL ENHIGH
        RET                 QMF DONE
*
*     SET UNVOICED STRATEGY AND CLEAR VOICING DECISION BUFFERS
SETUN   LACK 2
        SACL STRAT
        ZAC
        SACL ENLOW
        SACL ENHIGH
        RET                 QMF DONE
*
*                               QMF FILTERING DONE
*==============================================================================
```

APPENDIX D

Subroutine PCMAQB (coder)

```
*************************************************************************
*                            PCMAQB CODING                              *
*                                                                       *
* THIS SUBROUTINE INCORPORATES :                                        *
*                           -BACKWARD STEPSIZE ADAPTION OF THE QUANTIZER *
*                           -QUANTIZATION OF A SAMPLE OF THE SUBBAND    *
*                            BEING PROCESSED                            *
*                           -2,3 or 4 BIT PCM CODING OF THAT QUANTIZED  *
*                            SAMPLE                                     *
*                                                                       *
* INPUT:  FAW + 1 kHz SAMPLES (8 STREAMS)                               *
* OUTPUT: FAW + 1 kHz SAMPLES (8 STREAMS)                               *
*-----------------------------------------------------------------------*
* BIT ALLOCATION ACCORDING TO SSTRAT :                                  *
*                                                                       *
*                  ______________1__2__3__4__5__6__7__8____             *
*                  VOICED        4  4  3  2  2  0  0  0                 *
*                  INTERMEDIATE  4  3  2  2  2  2  0  0                 *
*                  UNVOICED      2  3  3  3  2  2  0  0                 *
*-----------------------------------------------------------------------*
* AUTHOR : ROLAND KLEUTERS                                              *
* DATE : 29-5-1987                                                      *
*************************************************************************
*
PCMAQB  NOP                 ENTRY POINT
*
*     READ IN FROM TABLE THE NUMBER OF CODE BITS FOR THE SUBBAND IN PROCESSION
        LACK BAVOI1
        ADD  PATH
        ADD  SSTRAT,3
        TBLR NBIT           NBIT CONTAINS NUMBER OF CODE BITS
*
*     READ IN THE LOWEST AND HIGHEST POSSIBLE INPUT VALUE OF THE EXPONENT TABLE
        LACK XX10
        ADD  PATH,1
        TBLR X0             X0 CONTAINS LOWEST POSSIBLE INPUT OF EXP. TABLE
        ADD  ONE
        TBLR X127           X127 CONTAINS HIGHEST POSSIBLE INPUT OF EXP. TABLE
*
*     CHECK IF NUMBER OF CODE BITS (NBIT) GREATER THAN ZERO
        LAC  NBIT
        BGZ  READIN         CODE?
*
*     NO CODING HAS TO TAKE PLACE (NBIT=0)
        ZAC                 ZERO "CODED" RESULT; I(n):=0
        SACL CALC1
        LACK COD1
        ADD  PATH
        TBLW CALC1          I(n) SAVED IN CODED SUBBAND SAMPLES TABLE
        LACK DN1            MINIMALIZE STEPSIZE; d(n):=X0
        ADD  PATH
        TBLW X0             d(n) SAVED IN TABLE
        RET
*
```

```
*--------------------------------------------------------------------------
* DETERMINE 1/STEPSIZE i.e. 1/DELTA(n) BY BACKWARD STEPSIZE ADAPTION
*--------------------------------------------------------------------------
*
*..........................................................................
*                          DETERMINE d(n)=LOG(DELTA(n))
*..........................................................................
*
*     READ IN THE PREVIOUS CODED RESULT I(n-1) FROM TABLE
READIN  LACK COD1
        ADD  PATH
        TBLR CALC1          CALC1 CONTAINS THE PREVIOUS CODED RESULT I(n-1)
*
*     DETERMINE  m([I(n-1)])=LOG(M([I(n-1)])) FROM TABLES
        LAC  NBIT
        SUB  ONE,1
        SACL CALC2
        LACK LOG22
        ADD  CALC2,4
        ADD  CALC1
        TBLR RES1           RES1 CONTAINS m([I(n-1)])=LOG(M[I(n-1)])
*
*     DETERMINE (1-GAMMA)*DC FROM TABLE
        LACK OGD1
        ADD  PATH
        TBLR CALC1          CALC1 CONTAINS (1-GAMMA)*DC
*
*     DETERMINE GAMMA*d(n-1) FROM TABLE
        LACK DN1
        ADD  PATH
        TBLR RES0           RES0 CONTAINS d(n-1)
        LT   GAMMA
        MPY  RES0
        PAC
        SACH CALC2,1        CALC2 CONTAINS GAMMA*d(n-1)
*
*     DETERMINE GAMMA*d(n-1)+m([I(n-1)])+(1-GAMMA)*DC
        ZALH CALC2
        ADD  RES1,12
        ADD  CALC1,8
        SACH CALC1          CALC1 CONTAINS GAMMA*d(n-1)+m([I(n-1)])+
*                           (1-GAMMA)*DC, i.e. d(n)
*
*     CONTROL THE LIMITATION OF d(n) TO ITS RANGE AND SAVE d(n) IN TABLE
        LAC  CALC1
        SUB  X127
        BGZ  SATDNH         d(n) GREATER THAN MAXIMUM POSSIBLE VALUE OF d(n)?
        LAC  X0
        SUB  CALC1
        BGZ  SATDNL         d(n) LESS THAN MINIMUM POSSIBLE VALUE OF d(n)?
        B    SAVEDN         NO SATURATION NEEDED
SATDNH  LAC  X127           SATURATE d(n) TO MAXIMUM POSSIBLE VALUE
        SACL CALC1
        B    SAVEDN
SATDNL  LAC  X0             SATURATE d(n) TO MINIMUM POSSIBLE VALUE
        SACL CALC1
```

```
SAVEDN  LACK DN1          SAVE d(n) IN TABLE
        ADD  PATH
        TBLW CALC1
*
*.......................................................................
*                   DETERMINE 1/DELTA(n)=EXP(-d(n))
*.......................................................................
*
*    ROUND d(n) TO ONE OF THE ENTRY POINTS OF THE EXPONENT TABLE
        ZALH CALC1
        SUBH X0
        SACH CALC1        CALC1 CONTAINS d(n)-X0
        LT   CALC1
        MPY  STEP
        PAC
        SACH CALC1        CALC1 CONTAINS (d(n)-X0)*STEP
        LAC  CALC1,11
        SACH CALC1        CALC1 CONTAINS INTEGER[(d(n)-X0)*STEP]=ENTRY POINT
*
*    DETERMINE THE OUTPUT OF EXPONENT TABLE
        LACK CEXPO
        ADD  CALC1
        TBLR CALC1        CALC1 CONTAINS OUTPUT OF EXPONENT TABLE=1/DELTA(n)
*
*-----------------------------------------------------------------------
* QUANTIZE THE PCMAQB CODER INPUT AND CODE THE RESULT (DETERMINE I(n))
*-----------------------------------------------------------------------
*
*    DETERMINE INPUT/DELTA(n) AND ROUND TO THE GREATEST INTEGER BELOW
        LT   CALC1
        MPY  INPCM
        PAC
        SACH CALC1        CALC1 CONTAINS INPUT/DELTA (Q18)
        LAC  CALC1,14
        SACH CALC1        CALC1 CONTAINS INTEGER[INPUT/DELTA] IN
*                         TWO's COMPLEMENT CODE, i.e. THE CODED RESULT I(n)
*
*    CONTROL THE LIMITATION OF I(n) TO THE RANGE ACCORDING TO THE NUMBER OF
*    CODE BITS AND SAVE I(n) IN TABLE OF CODED SUBBAND SAMPLES
        LACK SAT1
        ADD  NBIT
        SUB  ONE,1
        TBLR RES0         RES0 CONTAINS THE MAXIMUM POSSIBLE VALUE OF I(n)
        LAC  CALC1
        SUB  RES0
        BGZ  SATINH       I(n) GREATER THAN MAXIMUM POSSIBLE VALUE OF I(n)?
        ZAC
        SUB  RES0
        SUB  ONE
        SACL RES1         RES1 CONTAINS THE MINIMUM POSSIBLE VALUE OF I(n)
        SUB  CALC1
        BGZ  SATINL       I(n) LESS THAN MINIMUM POSSIBLE VALUE OF I(n)?
        LACK COD1         SAVE I(n) WITHOUT SATURATION
        ADD  PATH
        TBLW CALC1
        RET
```

```
SATINH  LACK COD1          SATURATE I(n) TO MAXIMUM POSSIBLE VALUE AND SAVE
        ADD  PATH
        TBLW RES0
        RET
SATINL  LACK COD1          SATURATE I(n) TO MINIMUM POSSIBLE VALUE AND SAVE
        ADD  PATH
        TBLW RES1
        RET
*
*                             PCMAQB CODING DONE
*=============================================================================
```

APPENDIX E

Subroutine MULTIPLEXING

```
****************************************************************************
*                           MULTIPLEXING                                  *
*                                                                         *
* THIS SUBROUTINE INCORPORATES:                                           *
*                          -MULTIPLEXING AND                              *
*                          -SENDING OF PCM CODED DATA                     *
*                                                                         *
* INPUT:  FAW + 1 kHz SAMPLES (8 STREAMS)                                 *
* OUTPUT: 16 KBPS BITSTREAM                                               *
*-------------------------------------------------------------------------*
* SUBBAND PROCESSING : 1,8,4,5,2,7,3,6                                    *
*                                                                         *
* BIT ALLOCATION :                                                        *
*                 ______________1__2__3__4__5__6__7__8____                *
*                 VOICED        4  4  3  2  2  0  0  0                    *
*                 INTERMEDIATE  4  3  2  2  2  2  0  0                    *
*                 UNVOICED      2  3  3  3  2  2  0  0                    *
* FAW's :                                                                 *
*        VOICED >76       INTERMEDIATE >96        UNVOICED >CC            *
*-------------------------------------------------------------------------*
* AUTHOR: ROLAND KLEUTERS                                                 *
* DATE: 29-5-1987                                                         *
****************************************************************************
*
*---------------------------------------------------------------------------
* MACRO
*---------------------------------------------------------------------------
*
*    MACRO TO SEND ONE BIT
SEND    $MACRO
        LAC  DATA,1
        SACL DATA            SENDBIT PLACED IN FRONT OF BITSELECTOR
        LAC  ONE,8
        AND  DATA            SELECT SENDBIT
        SACL SIGOUT          SENDBIT IS BIT 8 OF SIGOUT
        OUT  SIGOUT,3        SENDBIT SENDED VIA PIN 38 OF EXPANSION PORT
        $END
*
*---------------------------------------------------------------------------
* MACRO DEFINTION DONE; MAIN PART OF SUBROUTINE BEGINS
*---------------------------------------------------------------------------
*
MUX     NOP                  ENTRY POINT
*
*    SELECT FAW or ONE OF THE SUBBANDS FROM WHICH DATA HAS TO BE SENT
FAWSUB  LACK 4
        SUB  BAND
        BGZ  FWSBGZ
        BZ   NEXT4           SUBBAND 4 (1500-2000 Hz)
        ADD  ONE
        BZ   NEXT5           SUBBAND 5 (2000-2500 Hz)
        B    NEXT6           SUBBAND 6 (2500-3000 Hz)
```

```
FWSBGZ  SUB  ONE
        BZ   NEXT3          SUBBAND 3 (1000-1500 Hz)
        SUB  ONE
        BZ   NEXT2          SUBBAND 2 (500-1000 Hz)
        SUB  ONE
        BZ   NEXT1          SUBBAND 1 (0-500 Hz)
*                           FAW
*
*-------------------------------------------------------------------------------
* SEND FAW BITS
*-------------------------------------------------------------------------------
*
NFAW    ZALS DEMO           NEW ACCESS CHECK
        BZ   FSIG
*
*     INITIALIZATION FOR A NEW FAW ACCESS
        ZAC
        SACL DEMO           NO NEW ACCESS ANYMORE
        LACK 7
        SACL COUNT          8 FAW BITS TO SEND
*
*     LOAD FAW INTO SENDBUFFER (=DATA)
        ZALS SSTRAT
        BZ   SVOICE
        XOR  ONE
        BZ   SINTER
SUNVOI  LACK >CC            CODE FOR UNVOICED STRATEGY
        SACL DATA           PRESET UNVOICED STRATEGY
        B    FSIG
SVOICE  LACK >76            CODE FOR VOICED STRATEGY
        SACL DATA           PRESET VOICED STRATEGY
        B    FSIG
SINTER  LACK >96            CODE FOR INTERMEDIATE STRATEGY
        SACL DATA           PRESET INTERMEDIATE STRATEGY
*
*     SEND A FAW BIT
FSIG    SEND
        LAR  0,COUNT
        BANZ CONT
*
*     ALL FAW BITS SENT,PREPARE NEXT SUBBAND
        LACK >FF
        SACL DEMO           NEW ACCESS MARK
        LACK 1
        SACL BAND           NEXT SUBBAND TO SEND (0-500 Hz)
        RET
*
*     NOT ALL FAW BITS SEND,PREPARE NEXT BIT
CONT    SAR  0,COUNT
        RET
*
*-------------------------------------------------------------------------------
* SEND SUBBAND BITS
*-------------------------------------------------------------------------------
*
```

```
*.............................................................................
*                  A BIT OF SUBBAND 1 (0-500 Hz) HAS TO BE SENT
*.............................................................................
*
NEXT1   ZALS DEMO           NEW ACCESS CHECK
        BZ   LOOP1
*
*     INITIALIZATION FOR A NEW SUBBAND 1 ACCESS
        ZAC
        SACL DEMO           NO NEW ACCESS ANYMORE
        LACK COD1
        TBLR DATA           LOAD SUBBAND 1 DATA INTO SEND BUFFER
        LACK 2
        XOR  SSTRAT         UNVOICED?
        BNZ  VOIN1
*
*     FURTHER INITIALIZATION FOR UNVOICED STRATEGY
        LACK 1
        SACL COUNT          2 BITS TO SEND
        LAC  DATA,6         ADJUST DATA ACCORDING TO BITSELECTOR
        SACL DATA           DATA=0000 0000 DD00 0000
        B    LOOP1
*
*     FURTHER INITIALIZATION FOR VOICED AND INTERMEDIATE STRATEGY
VOIN1   LACK 3
        SACL COUNT          4 BITS TO SEND
        LAC  DATA,4         ADJUST DATA ACCORDING TO BITSELECTOR
        SACL DATA           DATA=0000 0000 DDDD 0000
*
*     SEND A SUBBAND 1 BIT
LOOP1   SEND
        LAR  0,COUNT
        BANZ CONT1
*
*     ALL SUBBAND 1 BITS SENT,PREPARE NEXT SUBBAND
        LACK >FF
        SACL DEMO           NEW ACCESS MARK
        LACK 4
        SACL BAND           NEXT SUBBAND TO SEND (1500-2000 Hz)
        RET
*
*     NOT ALL SUBBAND 1 BITS SENT,PREPARE NEXT BIT
CONT1   SAR  0,COUNT
        RET
*
*.............................................................................
*              A BIT OF SUBBAND 2 (500-1000 Hz) HAS TO BE SENT
*.............................................................................
*
NEXT2   ZALS DEMO           NEW ACCESS CHECK
        BZ   LOOP2
*
```

```
*     INITIALIZATION FOR A NEW SUBBAND 2 ACCESS
        ZAC
        SACL DEMO            NO NEW ACCESS ANYMORE
        LACK COD2
        TBLR DATA            LOAD SUBBAND 2 DATA INTO SEND BUFFER
        LAC  SSTRAT          VOICED?
        BNZ  INUN2
*
*     FURTHER INITIALIZATION FOR VOICED STRATEGY
        LACK 3
        SACL COUNT           4 BITS TO SEND
        LAC  DATA,4          ADJUST DATA ACCORDING TO BITSELECTOR
        SACL DATA            DATA=0000 0000 DDDD 0000
        B    LOOP2
*
*     FURTHER INITIALIZATION FOR INTERMEDIATE AND UNVOICED STRATEGY
INUN2   LACK 2
        SACL COUNT           3 BITS TO SEND
        LAC  DATA,5          ADJUST DATA ACCORDING TO BITSELECTOR
        SACL DATA            DATA=0000 0000 DDD0 0000
*
*     SEND A SUBBAND 2 BIT
LOOP2   SEND
        LAR  0,COUNT
        BANZ CONT2
*
*     ALL SUBBAND 2 BITS SENT,PREPARE NEXT SUBBAND
        LACK >FF
        SACL DEMO            NEW ACCESS MARK
        LACK 3
        SACL BAND            NEXT SUBBAND TO SEND (1000-1500 Hz)
        RET
*
*     NOT ALL SUBBAND 2 BITS SENT,PREPARE NEXT BIT
CONT2   SAR  0,COUNT
        RET
*
*..............................................................................
*                   A BIT OF SUBBAND 3 (1000-1500) HAS TO BE SENT
*..............................................................................
*
NEXT3   ZALS DEMO            NEW ACCESS CHECK
        BZ   LOOP3
*
*     INITIALIZATION FOR A NEW SUBBAND 3 ACCESS
        ZAC
        SACL DEMO            NO NEW ACCESS ANYMORE
        LACK COD3
        TBLR DATA            LOAD SUBBAND 3 DATA INTO SEND BUFFER
        LACK 1
        XOR  SSTRAT          INTERMEDIATE?
        BNZ  VOUN3
*
```

```
*     FURTHER INITIALIZATION FOR INTERMEDIATE STRATEGY
        LACK 1
        SACL COUNT          2 BITS TO SEND
        LAC  DATA,6         ADJUST DATA ACCORDING TO BITSELECTOR
        SACL DATA           DATA=0000 0000 DD00 0000
        B    LOOP3
*
*     FURTHER INITIALIZATION FOR VOICED AND UNVOICED STRATEGY
VOUN3   LACK 2
        SACL COUNT          3 BITS TO SEND
        LAC  DATA,5         ADJUST DATA ACCORDING TO BITSELECTOR
        SACL DATA           DATA=0000 0000 DDD0 0000
*
*     SEND A SUBBAND 3 BIT
LOOP3   SEND
        LAR  0,COUNT
        BANZ CONT3
*
*     ALL SUBBAND 3 BITS SENT AND PERHAPS ALSO END OF 15 BIT DATABLOCK
*     CHECK FOR END OF FRAME
        LACK >FF
        SACL DEMO           NEW ACCESS MARK
        LACK 7
        XOR  FRAME
        BZ   ENDFRM
*
*     NOT END OF FRAME,PREPARE NEXT SUBBAND
        LAC  SSTRAT         VOICED?
        BZ   NENDFM
        LACK 6
        SACL BAND           NEXT SUBBAND TO SEND (2500-3000 Hz)
        RET
*
*     NOT ALL SUBBAND 3 BITS SENT,PREPARE NEXT BIT
CONT3   SAR  0,COUNT
        RET
*
*...........................................................................
*                A BIT OF SUBBAND 4 (1500-2000 Hz) HAS TO BE SENT
*...........................................................................
*
NEXT4   ZALS DEMO           NEW ACCESS CHECK
        BZ   LOOP4
*
*     INITIALIZATION FOR A NEW SUBBAND 4 ACCESS
        ZAC
        SACL DEMO           NO NEW ACCESS ANYMORE
        LACK COD4
        TBLR DATA           LOAD SUBBAND 4 DATA INTO SEND BUFFER
        LACK 2
        XOR  SSTRAT         UNVOICED?
        BNZ  VOIN4
*
```

```
*     FURTHER INITIALIZATION FOR UNVOICED STRATEGY
        LACK 2
        SACL COUNT          3 BITS TO SEND
        LAC  DATA,5         ADJUST DATA ACCORDING TO BITSELECTOR
        SACL DATA           DATA=0000 0000 DDD0 0000
        B    LOOP4
*
*     FURTHER INITIALIZATION FOR VOICED AND INTERMEDIATE STRATEGY
VOIN4   LACK 1
        SACL COUNT          2 BITS TO SEND
        LAC  DATA,6         ADJUST DATA ACCORDING TO BITSELECTOR
        SACL DATA           DATA=0000 0000 DD00 0000
*
*     SEND A SUBBAND 4 BIT
LOOP4   SEND
        LAR  0,COUNT
        BANZ CONT4
*
*     ALL SUBBAND 4 BITS SENT PREPARE NEXT SUBBAND
        LACK >FF
        SACL DEMO           NEW ACCESS MARK
        LACK 5
        SACL BAND           NEXT SUBBAND TO SEND (2000-2500 Hz)
        RET
*
*     NOT ALL SUBBAND 4 BITS SENT,PREPARE NEXT BIT
CONT4   SAR  0,COUNT
        RET
*
*.............................................................................
*                A BIT OF SUBBAND 5 (2000-2500) HAS TO BE SENT
*.............................................................................
*
NEXT5   ZALS DEMO           NEW ACCESS CHECK
        BZ   LOOP5
*
*     INITIALIZATION FOR A NEW SUBBAND 5 ACCESS
        ZAC
        SACL DEMO           NO NEW ACCESS ANYMORE
        LACK COD5
        TBLR DATA           LOAD SUBBAND 5 DATA INTO SEND BUFFER
        LACK 1
        SACL COUNT          2 BITS TO SEND FOR ALL STRATEGIES
        LAC  DATA,6         ADJUST DATA ACCORDING TO BITSELECTOR
        SACL DATA           DATA=0000 0000 DD00 0000
*
*     SEND A SUBBAND 5 BIT
LOOP5   SEND
        LAR  0,COUNT
        BANZ CONT5
*
*     ALL SUBBAND 5 BITS SENT,PREPARE NEXT SUBBAND
        LACK >FF
        SACL DEMO           NEW ACCESS MARK
        LACK 2
        SACL BAND           NEXT SUBBAND TO SEND (500-1000 Hz)
        RET
*
```

```
*     NOT ALL SUBBAND 5 BITS SENT,PREPARE NEXT BIT
CONT5   SAR  0,COUNT
        RET
*
*.........................................................................
*               A BIT OF SUBBAND 6 (2500-3000 Hz) HAS TO BE SENT
*             ONLY ACCESSED  BY UNVOICED AND INTERMEDIATE STRATEGY
*.........................................................................
*
NEXT6   ZALS DEMO           NEW ACCESS CHECK
        BZ   LOOP6
*
*     INITIALIZATION FOR A NEW SUBBAND 6 ACCESS
        ZAC
        SACL DEMO           NO NEW ACCESS ANYMORE
        LACK COD6
        TBLR DATA           LOAD SUBBAND 6 DATA INTO SEND BUFFER
        LACK 1
        SACL COUNT          2 BITS TO SEND FOR ALL POSSIBLE STATEGIES
        LAC  DATA,6         ADJUST DATA ACCORDING TO BITSELECTOR
        SACL DATA           DATA=0000 0000 DD00 0000
*
*     SEND A SUBBAND 6 BIT
LOOP6   SEND
        LAR  0,COUNT
        BANZ CONT6
*
*     ALL SUBBAND 6 BITS SENT AND ALSO END OF 15 BIT DATABLOCK
*     CHECK FOR END OF FRAME
        LACK >FF
        SACL DEMO           NEW ACCESS MARK
        LACK 7
        XOR  FRAME
        BNZ  NENDFM
*
*     END OF FRAME,PREPARE NEXT FRAME
ENDFRM  ZAC
        SACL BAND           NEXT TO SEND=FAW
        RET
*
*     NOT END OF FRAME,PREPARE NEXT DATABLOCK
NENDFM  LACK 1
        SACL BAND           NEXT SUBBAND TO SEND (0-500 Hz)
        RET
*
*     NOT ALL SUBBAND 6 BITS SENT,PREPARE NEXT BIT
CONT6   SAR  0,COUNT
        RET
*
*                             MULTIPLEXING DONE
*=========================================================================
```

APPENDIX F

Main program SBC receiver part

```
*************************************************************************
*                          DECODER PROGRAM                              *
*                                                                       *
* THIS MAIN PROGRAM USES THE SUBROUTINES:                               *
*                                   -DEMULTIPLEXING                     *
*                                   -PCMAQB DECODING                    *
*                                   -INVQMF FILTERING                   *
* TO REALIZE A 16 kBIT PER SECOND SUBBAND DECODER.                      *
* SYSTEM CLOCK RATE: 16 kHz                                             *
*-----------------------------------------------------------------------*
* REPORTS FOR DETAILS:                                                  *
* PT REPORT BY P.CHITAMU                                                *
* FINAL REPORT BY J.BOLT.                                               *
* FINAL REPORT BY R.KLEUTERS                                            *
*-----------------------------------------------------------------------*
* AUTHOR: ROLAND KLEUTERS                                               *
* DATE: 29-5-1987                                                       *
*************************************************************************
*
*------------------------------------------------------------------------
* DATA MEMORY INITIALIZATION
*                                                         default page=0
*------------------------------------------------------------------------
*
*........................................................................
*                 GENERAL VARIABLES DATA MEMORY PAGE 1
*........................................................................
*
CLOCK   EQU  0          CLOCK RATE (16 kHz)
MODE    EQU  1          MODE FOR ANALOG INTERFACE BOARD
AR00    EQU  2          STORAGE PLACE FOR AUXILIARY REGISTER 0
AR01    EQU  3          STORAGE PLACE FOR AUXILIARY REGISTER 1
ACH     EQU  4          STORAGE PLACE FOR HIGH PART OF ACCUMULATOR
ACL     EQU  5          STORAGE PLACE FOR LOW PART OF ACCUMULATOR
TREG    EQU  6          STORAGE PLACE FOR T REGISTER
STATU   EQU  7          STORAGE PLACE FOR STATUS WORD OF TMS32010
*                                            from now on default page
*........................................................................
*                         GENERAL VARIABLES
*........................................................................
*
MSTAT   EQU  0          MASK FOR STATE MOD 16
SWITCH  EQU  1          >FF=SEARCH MODE,ELSE=RECEIVE MODE; 0=DECODER ON(Q0)
FRAME   EQU  2          COUNTER FOR FRAMELENGTH                       (Q0)
ONE     EQU  3          CHECK BIT                                     (Q0)
STRAT   EQU  4          CONTAINS 0, 1, or 2: VOICED, INTERM. or UNVO. (Q0)
SSTRAT  EQU  5          BUFFERING                                     (Q0)
STATE   EQU  6          TREE POINTER                                  (Q0)
RES0    EQU  7          INTERMEDIATE CALCULATION RESULTS FOR MAIN ROUTINE
RES1    EQU  8          INTERMEDIATE CALCULATION RESULTS FOR MAIN ROUTINE
PATH    EQU  9          CONTAINS STATE/2=CHANNEL TO PROCESS           (Q0)
OUTP    EQU  10         8 kHz OUTPUT SAMPLE OF TMS32010              (Q15)
BOUTP1  EQU  11         BUFFERED 8 kHz OUTPUT SAMPLE OF TMS32010     (Q15)
BOUTP2  EQU  12         16 kHz INTERPOLATION OF TMS32010 OUTPUT      (Q15)
*
```

```
*...........................................................................
*                         FIRST INVQMF STAGE VARIABLES
*...........................................................................
*
*     BUFFER VARIABLES OF LOW1 (0-2 kHz) AND HIGH1 (2-4 kHz) FOR.=Q14 RESP. Q16
A1      EQU  13
B1      EQU  14
C1      EQU  15
*
*     INPUT TO (FILTER PART OF) FIRST INVQMF STAGE                  FORMAT=Q15
INPF    EQU  16
*
*     DELAY VARIABLES OF LOW1 (0-2 kHz)                             FORMAT=Q15
XF2     EQU  17           AFTER FILTERING INPUT STORED IN XF2, SO
XF3     EQU  18           XF1 NOT NEEDED
XF4     EQU  19
XF5     EQU  20
XF6     EQU  21
XF7     EQU  22
XF8     EQU  23
XF9     EQU  24
XF10    EQU  25
XF11    EQU  26
XF12    EQU  27
XF13    EQU  28
XF14    EQU  29
XF15    EQU  30
XF16    EQU  31
*
*     DELAY VARIABLES OF HIGH1 (2-4 kHz)                            FORMAT=Q15
XF18    EQU  32           AFTER FILTERING INPUT STORED IN XF18, SO
XF19    EQU  33           XF17 NOT NEEDED
XF20    EQU  34
XF21    EQU  35
XF22    EQU  36
XF23    EQU  37
XF24    EQU  38
XF25    EQU  39
XF26    EQU  40
XF27    EQU  41
XF28    EQU  42
XF29    EQU  43
XF30    EQU  44
XF31    EQU  45
XF32    EQU  46
*
*...........................................................................
*                         SECOND INVQMF STAGE VARIABLES
*...........................................................................
*
*     BUFFER VARIABLES OF LOW2 (0-1 kHz) AND HIGH2 (1-2 kHz)        FORMAT=Q14
A2      EQU  47
B2      EQU  48
C2      EQU  49
*
```

```
*     BUFFER VARIABLES OF LOW3 (3-4 kHz) AND HIGH3 (2-3 kHz)        FORMAT=Q17
A3      EQU  50
B3      EQU  51
C3      EQU  52
*
*     INPUT TO (FILTER PART OF) SECOND INVQMF STAGE
INPS    EQU  53
*
*     DELAY VARIABLES OF LOW2,HIGH2,LOW3 AND HIGH3 (TIME SHARING)
XS2     EQU  54
XS3     EQU  55
XS4     EQU  56
XS5     EQU  57
XS6     EQU  58
XS7     EQU  59
XS8     EQU  60
*
*..........................................................................
*                         THIRD INVQMF STAGE VARIABLES
*..........................................................................
*
*     BUFFER VARIABLES OF LOW4 (0-500 Hz) AND HIGH4 (500-1000 Hz)    FORMAT=Q14
A4      EQU  61
B4      EQU  62
C4      EQU  63
*
*     BUFFER VARIABLES OF LOW5 (1500-2000 Hz) AND HIGH5 (1000-1500 Hz) FOR.=Q15
A5      EQU  64
B5      EQU  65
C5      EQU  66
*
*     BUFFER VARIABLES OF LOW7 (2000-2500 Hz) AND HIGH7 (2500-3000 Hz) FOR.=Q18
A7      EQU  67
B7      EQU  68
C7      EQU  69
*
*     INPUT TO (FILTER PART OF) THIRD INVQMF STAGE
INPT    EQU  70
*
*     DELAY VARIABLES OF LOW4,HIGH4,LOW5,HIGH5,LOW7 AND HIGH7 (TIME SHARING)
XT2     EQU  71
XT3     EQU  72
XT4     EQU  73
XT5     EQU  74
XT6     EQU  75
*
*..........................................................................
*                        FIRST INVQMF STAGE COEFFICIENTS            FORMAT=Q16
*..........................................................................
*
CF0     EQU  76
CF1     EQU  77
CF2     EQU  78
CF3     EQU  79
CF4     EQU  80
CF5     EQU  81
CF6     EQU  82
CF7     EQU  83
```

```
CF8     EQU  84
CF9     EQU  85
CF10    EQU  86
CF11    EQU  87
CF12    EQU  88
CF13    EQU  89
CF14    EQU  90
CF15    EQU  91
*
*.........................................................................
*                        SECOND INVQMF STAGE COEFFICIENTS          FORMAT=Q16
*.........................................................................
*
CS0     EQU  92
CS1     EQU  93
CS2     EQU  94
CS3     EQU  95
CS4     EQU  96
CS5     EQU  97
CS6     EQU  98
CS7     EQU  99
*
*.........................................................................
*                        THIRD INVQMF STAGE COEFFICIENTS           FORMAT=Q16
*.........................................................................
*
CT0     EQU  100
CT1     EQU  101
CT2     EQU  102
CT3     EQU  103
CT4     EQU  104
CT5     EQU  105
*
*.........................................................................
*                               PCMAQB VARIABLES
*.........................................................................
*
OUTPCM  EQU  106           OUTPUT OF PCMAQB DECODER
CALC1   EQU  107           GENERAL VARIABLE TO CALCULATE WITH
CALC2   EQU  108           GENERAL VARIABLE TO CALCULATE WITH
NBIT    EQU  109           CONTAINS NR. OF BITS TO QUANTIZE TO          (Q0)
X0      EQU  110           LOWEST POSSIBLE INPUT VALUE OF EXPONENT TABLE (Q14)
X127    EQU  111           HIGHEST POSSIBLE INPUT VALUE OF EXPONENT TABLE(Q14)
STEP    EQU  112           1/(DISTANCE BETWEEN INPUTS OF EXPONENT TABLE)  (Q7)
GAMMA   EQU  113           GAMMA VALUE OF FORMULA (0.99)                 (Q15)
V3FFF   EQU  114           >3FFF; HELPCONSTANT
*
*.........................................................................
*                           DEMULTIPLEXER VARIABLES
*.........................................................................
*
POINT   EQU  115           FLAG: >FF=SEARCH FAW, 0=SKIP 128 BITS
ERROR   EQU  116           CONTAINS NUMBER OF ERROR ALLOWANCES            (Q0)
COUNT   EQU  117           MULTIPURPOSE COUNTER                           (Q0)
BAND    EQU  118           FLAG, CURRENT SUBBAND PROCESSING               (Q0)
DATA    EQU  119           STORES INPUT BITS VIA SIGIN
SIGIN   EQU  120           VARIABLE TO READ IN SERIAL DATA FROM OUTSIDE
DEMO    EQU  121           NEW (FAW or SUBBAND) ACCESS MARK
```

```
*
*-------------------------------------------------------------------------
* PROGRAM MEMORY INITIALIZATION
*-------------------------------------------------------------------------
*
        AORG 0
        B    START            BRANCH TO MAIN ROUTINE
        B    INTRO            BRANCH TO INTERRUPT SERVICE ROUTINE
        AORG 8                OTHERWISE ERRORS WITH TBLW'S
*
*.........................................................................
*                              GENERAL CONSTANTS
*.........................................................................
*
CCLOCK  DATA  312             CLOCK RATE (16 kHz)
MMODE   DATA  7               MODE FOR ANALOG INTERFACE BOARD
MMSTAT  DATA >000F            MASK FOR STATE MOD 16
*
*.........................................................................
*                        SECOND INVQMF STAGE VARIABLES
*.........................................................................
*
*     DELAY VARIABLES OF LOW2 (0-1 kHz)                            FORMAT=Q14
XXS2    DATA 0                AFTER FILTERING INPUT STORED IN XXS2, SO
XXS3    DATA 0                XXS1 NOT NEEDED
XXS4    DATA 0
XXS5    DATA 0
XXS6    DATA 0
XXS7    DATA 0
XXS8    DATA 0
*
*     DELAY VARIABLES OF HIGH2 (1-2 kHz)                           FORMAT=Q14
XXS10   DATA 0                AFTER FILTERING INPUT STORED IN XXS10, SO
XXS11   DATA 0                XXS9 NOT NEEDED
XXS12   DATA 0
XXS13   DATA 0
XXS14   DATA 0
XXS15   DATA 0
XXS16   DATA 0
*
*     DELAY VARIABLES OF LOW3 (3-4 kHz)                            FORMAT=Q16
XXS18   DATA 0                AFTER FILTERING INPUT STORED IN XXS18, SO
XXS19   DATA 0                XXS17 NOT NEEDED
XXS20   DATA 0
XXS21   DATA 0
XXS22   DATA 0
XXS23   DATA 0
XXS24   DATA 0
*
*     DELAY VARIABLES OF HIGH3 (2-3 kHz)                           FORMAT=Q16
XXS26   DATA 0                AFTER FILTERING INPUT STORED IN XXS26, SO
XXS27   DATA 0                XXS25 NOT NEEDED
XXS28   DATA 0
XXS29   DATA 0
XXS30   DATA 0
XXS31   DATA 0
XXS32   DATA 0
*
```

```
*.........................................................................
*                        THIRD INVQMF STAGE VARIABLES
*.........................................................................
*
*     DELAY VARIABLES OF LOW4 (0-500 Hz)                              FORMAT=Q14
XXT2   DATA 0              AFTER FILTERING INPUT STORED IN XXT2, SO
XXT3   DATA 0              XXT1 NOT NEEDED
XXT4   DATA 0
XXT5   DATA 0
XXT6   DATA 0
*
*     DELAY VARIABLES OF HIGH4 (500-1000 Hz)                          FORMAT=Q14
XXT8   DATA 0              AFTER FILTERING INPUT STORED IN XXT8, SO
XXT9   DATA 0              XXT7 NOT NEEDED
XXT10  DATA 0
XXT11  DATA 0
XXT12  DATA 0
*
*     DELAY VARIABLES OF LOW5 (1500-2000 Hz)                          FORMAT=Q14
XXT14  DATA 0              AFTER FILTERING INPUT STORED IN XXT14, SO
XXT15  DATA 0              XXT13 NOT NEEDED
XXT16  DATA 0
XXT17  DATA 0
XXT18  DATA 0
*
*     DELAY VARIABLES OF HIGH5 (1000-1500 Hz)                         FORMAT=Q14
XXT20  DATA 0              AFTER FILTERING INPUT STORED IN XXT20, SO
XXT21  DATA 0              XXT19 NOT NEEDED
XXT22  DATA 0
XXT23  DATA 0
XXT24  DATA 0
*
*     DELAY VARIABLES OF LOW7 (2000-2500 Hz)                          FORMAT=Q17
XXT26  DATA 0              AFTER FILTERING INPUT STORED IN XXT26, SO
XXT27  DATA 0              XXT25 NOT NEEDED
XXT28  DATA 0
XXT29  DATA 0
XXT30  DATA 0
*
*     DELAY VARIABLES OF HIGH7 (2500-3000 Hz)                         FORMAT=Q17
XXT32  DATA 0              AFTER FILTERING INPUT STORED IN XXT32, SO
XXT33  DATA 0              XXT31 NOT NEEDED
XXT34  DATA 0
XXT35  DATA 0
XXT36  DATA 0
*
*.........................................................................
*                      FIRST INVQMF STAGE COEFFICIENTS                FORMAT=Q16
*.........................................................................
*
CCF0   DATA  45
CCF1   DATA -92
CCF2   DATA -83
CCF3   DATA  277
CCF4   DATA  93
CCF5   DATA -620
CCF6   DATA -9
CCF7   DATA  1178
```

```
CCF8    DATA -274
CCF9    DATA -2047
CCF10   DATA  955
CCF11   DATA  3470
CCF12   DATA -2579
CCF13   DATA -6541
CCF14   DATA  8425
CCF15   DATA  30566
*
*..........................................................................
*                     SECOND INVQMF STAGE COEFFICIENTS            FORMAT=Q16
*..........................................................................
*
CCS0    DATA  69
CCS1    DATA -331
CCS2    DATA -170
CCS3    DATA  1812
CCS4    DATA -633
CCS5    DATA -5924
CCS6    DATA  6409
CCS7    DATA  31525
*
*..........................................................................
*                      THIRD INVQMF STAGE COEFFICIENTS            FORMAT=Q16
*..........................................................................
*
CCT0    DATA -250
CCT1    DATA  1236
CCT2    DATA -178
CCT3    DATA -5551
CCT4    DATA  5798
CCT5    DATA  31745
*
*..........................................................................
*                              PCMAQB's TABLES
*..........................................................................
*
*     (1-GAMMA)*DC FOR 7 SUBBANDS                                 FORMAT=Q22
OGD1    DATA -10486          SUBBAND 1
DUM1    DATA  0
OGD4    DATA -13642          SUBBAND 4
OGD5    DATA -23112          SUBBAND 5
OGD2    DATA -10486          SUBBAND 2
DUM2    DATA  0
OGD3    DATA -13642          SUBBAND 3
OGD6    DATA -23112          SUBBAND 6
*
*     D(N) FOR 7 SUBBANDS                                         FORMAT=Q14
DN1     DATA -13563          SUBBAND 1
DUM3    DATA  0
DN4     DATA -14796          SUBBAND 4
DN5     DATA -18495          SUBBAND 5
DN2     DATA -13563          SUBBAND 2
DUM4    DATA  0
DN3     DATA -14796          SUBBAND 3
DN6     DATA -18495          SUBBAND 6
*
```

```
*     BIT ALLOCATION FOR VOICED STRATEGY (0)                        FORMAT=Q0
BAVOI1  DATA 4              SUBBAND 1
DUM5    DATA 0
BAVOI4  DATA 2              SUBBAND 4
BAVOI5  DATA 2              SUBBAND 5
BAVOI2  DATA 4              SUBBAND 2
DUM6    DATA 0
BAVOI3  DATA 3              SUBBAND 3
BAVOI6  DATA 0              SUBBAND 6
*
*     BIT ALLOCATION FOR INTERMEDIATE STRATEGY (1)                  FORMAT=Q0
BAINT1  DATA 4              SUBBAND 1
DUM7    DATA 0
BAINT4  DATA 2              SUBBAND 4
BAINT5  DATA 2              SUBBAND 5
BAINT2  DATA 3              SUBBAND 2
DUM8    DATA 0
BAINT3  DATA 2              SUBBAND 3
BAINT6  DATA 2              SUBBAND 6
*
*     BIT ALLOCATION FOR UNVOICED STRATEGY (2)                      FORMAT=Q0
BAUNV1  DATA 2              SUBBAND 1
DUM9    DATA 0
BAUNV4  DATA 3              SUBBAND 4
BAUNV5  DATA 2              SUBBAND 5
BAUNV2  DATA 3              SUBBAND 2
DUM10   DATA 0
BAUNV3  DATA 3              SUBBAND 3
BAUNV6  DATA 2              SUBBAND 6
*
*     STORAGE OF CODED SUBBAND SAMPLES (INPUT PCMAQB DECODER)       FORMAT=Q0
COD1    DATA 0              SUBBAND 1
DUM11   DATA 0
COD4    DATA 0              SUBBAND 4
COD5    DATA 0              SUBBAND 5
COD2    DATA 0              SUBBAND 2
DUM12   DATA 0
COD3    DATA 0              SUBBAND 3
COD6    DATA 0              SUBBAND 6
*
*     STORAGE OF PREVIOUS CODED SUBBAND SAMPLES (STEPSIZE ADAPTION) FORMAT=Q0
PCOD1   DATA 0              SUBBAND 1
DUM13   DATA 0
PCOD4   DATA 0              SUBBAND 4
PCOD5   DATA 0              SUBBAND 5
PCOD2   DATA 0              SUBBAND 2
DUM14   DATA 0
PCOD3   DATA 0              SUBBAND 3
PCOD6   DATA 0              SUBBAND 6
*
*     QUANTIZER CONSTANTS
SSTEP   DATA  19637         1/(DISTANCE BETWEEN INPUTS OF EXPONENT TABLE) (Q7)
GGAMMA  DATA  32440         GAMMA=0.99                                   (Q15)
VV3FFF  DATA >3FFF          >3FFF; HELPCONSTANT
*
```

```
*     LOGTABLE 2 BIT CASE                                            FORMAT=Q18
LOG20   DATA  18268
LOG21   DATA -4626
LOG22   DATA -4626
LOG23   DATA  18268
DUM15   DATA  0
DUM16   DATA  0
DUM17   DATA  0
DUM18   DATA  0
DUM19   DATA  0
DUM20   DATA  0
DUM21   DATA  0
DUM22   DATA  0
DUM23   DATA  0
DUM24   DATA  0
*
*     LOGTABLE 3 BIT CASE                                            FORMAT=Q18
LOG30   DATA  11540
LOG31   DATA  0
LOG32   DATA  0
LOG33   DATA -4626
LOG34   DATA -4626
LOG35   DATA  0
LOG36   DATA  0
LOG37   DATA  11540
DUM25   DATA  0
DUM26   DATA  0
DUM27   DATA  0
DUM28   DATA  0
*
*     LOGTABLE 4 BIT CASE                                            FORMAT=Q18
LOG40   DATA  24918
LOG41   DATA  19728
LOG42   DATA  13377
LOG43   DATA  5189
LOG44   DATA -2999
LOG45   DATA -2999
LOG46   DATA -2999
LOG47   DATA -2999
LOG48   DATA -2999
LOG49   DATA -2999
LOG410  DATA -2999
LOG411  DATA -2999
LOG412  DATA  5189
LOG413  DATA  13377
LOG414  DATA  19728
LOG415  DATA  24918
*
*     LOWEST AND HIGHEST POSSIBLE INPUTS OF EXPONENT TABLE           FORMAT=Q14
XX10    DATA -13563          SUBBAND 1
XX1127  DATA  0              SUBBAND 1
DUM29   DATA  0
DUM30   DATA  0
XX40    DATA -14796          SUBBAND 4
XX4127  DATA -1233           SUBBAND 4
XX50    DATA -18495          SUBBAND 5
XX5127  DATA -4932           SUBBAND 5
XX20    DATA -13563          SUBBAND 2
```

```
XX2127  DATA  0           SUBBAND 2
DUM31   DATA  0
DUM32   DATA  0
XX30    DATA -14796       SUBBAND 3
XX3127  DATA -1233        SUBBAND 3
XX60    DATA -18495       SUBBAND 6
XX6127  DATA -4932        SUBBAND 6
*
*     EXPONENT TABLE OF DECODER
*     FORMATS ARE : SUBBAND 1  Q15
*                   SUBBAND 2  Q15
*                   SUBBAND 3  Q16
*                   SUBBAND 4  Q16
*                   SUBBAND 5  Q19
*                   SUBBAND 6  Q19
*
CEXPO   DATA 16
        DATA 17
        DATA 18
        DATA 19
        DATA 20
        DATA 22
        DATA 23
        DATA 24
        DATA 26
        DATA 27
        DATA 29
        DATA 31
        DATA 33
        DATA 35
        DATA 37
        DATA 39
        DATA 42
        DATA 44
        DATA 47
        DATA 50
        DATA 53
        DATA 56
        DATA 60
        DATA 64
        DATA 68
        DATA 72
        DATA 76
        DATA 81
        DATA 86
        DATA 91
        DATA 97
        DATA 103
        DATA 109
        DATA 116
        DATA 123
        DATA 131
        DATA 139
        DATA 148
        DATA 157
        DATA 166
        DATA 177
        DATA 188
```

```
DATA 199
DATA 211
DATA 225
DATA 238
DATA 253
DATA 269
DATA 286
DATA 303
DATA 322
DATA 342
DATA 363
DATA 385
DATA 409
DATA 435
DATA 462
DATA 490
DATA 520
DATA 553
DATA 587
DATA 623
DATA 662
DATA 703
DATA 746
DATA 792
DATA 841
DATA 893
DATA 949
DATA 1007
DATA 1070
DATA 1136
DATA 1206
DATA 1281
DATA 1360
DATA 1444
DATA 1534
DATA 1628
DATA 1729
DATA 1836
DATA 1950
DATA 2070
DATA 2199
DATA 2335
DATA 2479
DATA 2632
DATA 2795
DATA 2968
DATA 3152
DATA 3347
DATA 3554
DATA 3774
DATA 4008
DATA 4256
DATA 4519
DATA 4798
DATA 5095
DATA 5411
DATA 5745
DATA 6101
```

```
        DATA 6478
        DATA 6879
        DATA 7305
        DATA 7757
        DATA 8237
        DATA 8746
        DATA 9288
        DATA 9862
        DATA 10473
        DATA 11121
        DATA 11809
        DATA 12539
        DATA 13315
        DATA 14139
        DATA 15014
        DATA 15943
        DATA 16929
        DATA 17977
        DATA 19089
        DATA 20270
        DATA 21525
        DATA 22856
        DATA 24271
        DATA 25772
        DATA 27367
        DATA 29060
        DATA 30859
        DATA 32767
*
*-------------------------------------------------------------------------------
* MAIN ROUTINE
*-------------------------------------------------------------------------------
*
*...............................................................................
*                                  INITIALIZATIONS
*...............................................................................
*
START   DINT
*
*      INITIALIZE STATUS BITS
        LARP 0
        LDPK 0
        SOVM
*
*      INITIALIZE GENERAL VARIABLES OF DATA MEMORY PAGE 1
        LDPK 1
        LACK CCLOCK
        TBLR CLOCK          CLOCK RATE (16 kHz)
        LACK MMODE
        TBLR MODE           MODE FOR ANALOG INTERFACE BOARD
        LDPK 0
*
*      INITIALIZE GENERAL VARIABLES OF DATA MEMORY PAGE 0
        LACK MMSTAT
        TBLR MSTAT          MASK FOR STATE MOD 16
        LACK >FF
        SACL SWITCH         DEMULTIPLEXER IN SEARCH MODE; DECODER OFF
        ZAC
```

```
        SACL FRAME          STATUS FOR DECODER
        LACK 1
        SACL ONE            '1' FOR LOGICAL MANIPULATIONS
        SACL STRAT          BY DEFAULT LOAD INTERMEDIATE STRATEGY
        SACL SSTRAT         ALSO INTERMEDIATE STRATEGY FOR BUFFER
        ZAC
        SACL STATE          INIT TREE POINTER
        SACL OUTP           ZERO 8 kHz OUTPUT SAMPLE OF TMS32010
        SACL BOUTP1         ZERO BUFFERED 8 kHz OUTPUT SAMPLE OF TMS32010
        SACL BOUTP2         ZERO 16 kHz INTERPOLATED OUTPUT OF TMS32010
*
*      INITIALIZE PCMAQB VARIABLES
        LACK SSTEP
        TBLR STEP           1/(DISTANCE BETWEEN INPUTS OF EXPONENT TABLE)
        LACK GGAMMA
        TBLR GAMMA          GAMMA VALUE OF FORMULA (0.99)
        LACK VV3FFF
        TBLR V3FFF          >3FFF; HELPCONSTANT
*
*      INITIALIZE DEMULTIPLEXER VARIABLES
        LACK >FF
        SACL POINT          SEARCH FIRST FAW
*
*      LOAD INVQMF COEFFICIENTS INTO DATA RAM
        LARK 1,CT5
        LARK 0,29
        LACK CCT5
LOAD    LARP 1
        TBLR *-,0
        SUB  ONE
        BANZ LOAD
*
*      INITIALIZE ANALOG INTERFACE BOARD (AIB)
        LDPK 1
        OUT  MODE,0         PROGRAM AIB
        OUT  CLOCK,1        SET CLOCK RATE OF AIB
        LDPK 0
        OUT  OUTP,2         CLEAR AD1S (PROTECTION AGAINST RESET INTERRUPTS)
        OUT  OUTP,3         CLEAR AD2S (PROTECTION AGAINST RESET INTERRUPTS)
*
*      INITIALIZATION AND VARIABLE LOADING DONE, NOW BEGIN
        EINT
*
*..........................................................................
*                                    AORTA
*..........................................................................
*
*      ODD CYCLE OF 16 kHz CLOCK MUST JUST HAVE PASSED
WAITEV  LAC  STATE
        AND  ONE
        BNZ  WAITEV
*      EVEN CYCLE OF 16 kHz CLOCK HAS PASSED (BIT 0 OF STATE=0)
WAIT    LAC  STATE
        AND  ONE
        BZ   WAIT
*      ODD CYCLE OF 16 kHz CLOCK HAS JUST PASSED (BIT 0 OF STATE=1)
*
```

```
*     DURING THE PROCESSING OF A SUBBAND SAMPLE STATE CAN CHANGE AS A
*     RESULT OF AN INTERRUPT, SO DEFINE PATH=STATE/2
       LAC  STATE,15
       SACH PATH
*
*     CHECK IF DECODER IS ON (i.e. IN SYNCHRONIZATION)
       ZALS SWITCH
       BZ   ACTSUB         SWITCH=0?
*
*     DECODER IS NOT ON; OUTPUT A ZERO AND WAIT FOR NEXT CYCLE
       ZAC
       SACL OUTP
       B    WAITEV
*
*     DECODER IS ON; ACTUAL SUBBAND DECODING
ACTSUB CALL PCMAQB
       CALL INVQMF
       B    WAITEV
*
*------------------------------------------------------------------------------
* INTERRUPT SERVICE ROUTINE
*------------------------------------------------------------------------------
*
INTRO  DINT
*
*     SAVE STATUS OF TMS32010
       SST  STATU
       MPY  ONE            T REGISTER TO P REGISTER
       LDPK 1
       SAR  0,AR00
       SAR  1,AR01
       SACH ACH
       SACL ACL
       PAC
       SACL TREG
       LDPK 0
*
*     STATE UPDATE
       LAC  STATE
       ADD  ONE
       AND  MSTAT          STATE MOD 16
       SACL STATE
*
*     RUN THROUGH DEMULTIPLEXER
       CALL DEMUX
*
*     CREATE A 16 kHz OUTPUT SAMPLE OF THE TMS32010
*     OUTPUT IS 8 kHz INVQMF TREE OUTPUT IF CYCLE IS ODD
*     OUTPUT IS INTERPOLATION OF 8 kHz INVQMF TREE OUTPUT IF CYCLE IS EVEN
       LAC  STATE
       AND  ONE
       BZ   INTPOL
       LAC  OUTP           CYCLE IS ODD; OUTPUT IS INVQMF OUTPUT
       SACL BOUTP1
       OUT  BOUTP2,2
       B    RSSTAT
```

```
INTPOL  LAC  BOUTP1,15      CYCLE IS EVEN; OUTPUT IS INTERPOLATION
        ADD  BOUTP2,15
        SACH BOUTP2
        OUT  BOUTP2,2
        DMOV BOUTP1
*
*     RESTORE STATUS OF TMS32010
RSSTAT  LDPK 1
        LAR  0,AR00
        LAR  1,AR01
        ZALH ACH
        ADDS ACL
        LT   TREG
        LST  STATU
*
*     RETURN TO PLACE OF INTERRUPT AND CONTINUE
        EINT
        RET
*
*                              DECODER PROGRAM DONE
*==============================================================================
```

APPENDIX G

Subroutine PCMAQB (decoder)

```
******************************************************************************
*                         PCMAQB DECODING                                    *
*                                                                            *
* THIS SUBROUTINE INCORPORATES:                                              *
*                         -BACKWARD STEPSIZE ADAPTION OF THE QUANTIZER       *
*                         -2,3 or 4 BIT PCM DECODING OF A CODED SAMPLE       *
*                          INTO A QUANTIZED SAMPLE OF THE SUBBAND BEING      *
*                          PROCESSED                                         *
*                                                                            *
* INPUT:  FAW + 1 kHz SAMPLES (8 STREAMS)                                    *
* OUTPUT: 1 kHz SAMPLES (8 STREAMS)                                          *
*----------------------------------------------------------------------------*
* BIT ALLOCATION ACCORDING TO SSTRAT :                                       *
*                                                                            *
*                 ______________1__2__3__4__5__6__7__8___                    *
*                 VOICED        4  4  3  2  2  0  0  0                       *
*                 INTERMEDIATE  4  3  2  2  2  2  0  0                       *
*                 UNVOICED      2  3  3  3  2  2  0  0                       *
*----------------------------------------------------------------------------*
* AUTHOR : ROLAND KLEUTERS                                                   *
* DATE : 29-5-1987                                                           *
******************************************************************************
*
PCMAQB  NOP                 ENTRY POINT
*
*     READ IN FROM TABLE THE NUMBER OF CODE BITS FOR THE SUBBAND IN PROCESSION
        LACK BAVOI1
        ADD  PATH
        ADD  SSTRAT,3
        TBLR NBIT           NBIT CONTAINS NUMBER OF CODE BITS
*
*     READ IN THE LOWEST AND HIGHEST POSSIBLE INPUT VALUE OF THE EXPONENT TABLE
        LACK XX10
        ADD  PATH,1
        TBLR X0             X0 CONTAINS LOWEST POSSIBLE INPUT OF EXP. TABLE
        ADD  ONE
        TBLR X127           X127 CONTAINS HIGHEST POSSIBLE INPUT OF EXP. TABLE
*
*     CHECK IF NUMBER OF CODE BITS (NBIT) GREATER THAN ZERO
        LAC  NBIT
        BGZ  READIN         DECODE?
*
*     NO DECODING HAS TO TAKE PLACE (NBIT=0)
        ZAC                 ZERO "CODED" RESULT; I(n):=0
        SACL CALC1
        LACK PCOD1
        ADD  PATH
        TBLW CALC1          I(n) SAVED IN PREVIOUS CODED SUBBAND SAMPLES TABLE
        LACK DN1            MINIMALIZE STEPSIZE; d(n):=X0
        ADD  PATH
        TBLW X0             d(n) SAVED IN TABLE
        ZAC                 ZERO "DECODED" RESULT; QUANTIZED SUBBAND SAMPLE:=0
        SACL OUTPCM         QUANTIZED SUBBAND SAMPLE SAVED IN OUTPCM
        RET
*
```

```
*---------------------------------------------------------------------------
* DETERMINE STEPSIZE i.e. DELTA(N) BY BACKWARD STEPSIZE ADAPTION
*---------------------------------------------------------------------------
*
*...........................................................................
*                         DETERMINE d(n)=LOG(DELTA(n))
*...........................................................................
*
*    READ IN THE PREVIOUS CODED RESULT RESULT I(n-1) FROM TABLE
READIN  LACK PCOD1
        ADD  PATH
        TBLR CALC1          CALC1 CONTAINS THE PREVIOUS CODED RESULT I(n-1)
*
*    DETERMINE m([I(n-1)])=LOG(M([I(n-1)])) FROM TABLES
        LAC  NBIT
        SUB  ONE,1
        SACL CALC2
        LACK LOG22
        ADD  CALC2,4
        ADD  CALC1
        TBLR RES1           RES1 CONTAINS m([I(n-1)])=LOG(M([I(n-1)]))
*
*    DETERMINE (1-GAMMA)*DC FROM TABLE
        LACK OGD1
        ADD  PATH
        TBLR CALC1          CALC1 CONTAINS (1-GAMMA)*DC
*
*    DETERMINE GAMMA*d(n-1) FROM TABLE
        LACK DN1
        ADD  PATH
        TBLR RES0           RES0 CONTAINS d(n-1)
        LT   GAMMA
        MPY  RES0
        PAC
        SACH CALC2,1        CALC2 CONTAINS GAMMA*d(n-1)
*
*    DETERMINE GAMMA*d(n-1)+m([I(n-1)])+(1-GAMMA)*DC
        ZALH CALC2
        ADD  RES1,12
        ADD  CALC1,8
        SACH CALC1          CALC1 CONTAINS GAMMA*d(n-1)+m([I(n-1)])+
*                           (1-GAMMA)*DC, i.e d(n)
*
*    CONTROL THE LIMITATION OF d(n) TO ITS RANGE AND SAVE d(n) IN TABLE
        LAC  CALC1
        SUB  X127
        BGZ  SATDNH         d(n) GREATER THAN MAXIMUM POSSIBLE VALUE OF d(n)?
        LAC  X0
        SUB  CALC1
        BGZ  SATDNL         d(n) LESS THAN MINIMUM POSSIBLE VALUE OF d(n)?
        B    SAVEDN         NO SATURATION NEEDED
SATDNH  LAC  X127           SATURATE d(n) TO MAXIMUM POSSIBLE VALUE
        SACL CALC1
        B    SAVEDN
SATDNL  LAC  X0             SATURATE d(n) TO MINIMUM POSSIBLE VALUE
        SACL CALC1
```

```
SAVEDN  LACK DN1           SAVE d(n) IN TABLE
        ADD  PATH
        TBLW CALC1
*
*...............................................................................
*                          DETERMINE DELTA(n)=EXP(d(n))
*...............................................................................
*
*     ROUND d(n) TO ONE OF THE ENTRY POINTS OF THE EXPONENT TABLE
        ZALH CALC1
        SUBH X0
        SACH CALC1         CALC1 CONTAINS d(n)-X0
        LT   CALC1
        MPY  STEP
        PAC
        SACH CALC1         CALC1 CONTAINS (d(n)-X0)*STEP
        LAC  CALC1,11
        SACH CALC1         CALC1 CONTAINS INTEGER[(d(n)-X0)*STEP]=ENTRY POINT
*
*     DETERMINE THE OUTPUT OF THE EXPONENT TABLE
        LACK CEXPO
        ADD  CALC1
        TBLR CALC1         CALC1 CONTAINS OUTPUT OF EXPONENT TABLE=DELTA(n)
*
*-------------------------------------------------------------------------------
* DECODE PCMAQB DECODER INPUT (I(n)) INTO A QUANTIZED SUBBAND SAMPLE
*-------------------------------------------------------------------------------
*
*     READ IN I(n) FROM TABLE OF CODED SUBBAND SAMPLES AND
*     SAVE I(n) IN TABLE OF PREVIOUS CODED SUBBAND SAMPLES
        LACK COD1
        ADD  PATH
        TBLR CALC2         CALC2 CONTAINS CODED RESULT I(n)
        LACK PCOD1
        ADD  PATH
        TBLW CALC2         I(n) SAVED IN TABLE
*
*     DETERMINE (I(n)+0.5)*DELTA(n)
        LAC  CALC2,1
        ADD  ONE
        SACL CALC2         CALC2 CONTAINS I(n)+0.5 (FORMAT=Q1)
        LT   CALC2
        MPY  CALC1
        PAC                ACCU CONTAINS (I(n)+0.5)*DELTA(n), i.e. THE
*                          QUANTIZED SUBBAND SAMPLE (FORMAT=Q16,Q17 or Q20)
*
*     CONTROL THE LIMITATION OF THE QUANTIZED SUBBAND SAMPLE TO ITS RANGE AND
*     SAVE THE QUANTIZED SUBBAND SAMPLE (IN Q14,Q15 or Q18 FORMAT) IN OUTPCM
        SACL RES0          RES0=DDDD DDDD DDDD DD-- ; D=DATA BIT -=DON'T CARE
        SACH RES1          RES1=SSSS SSSS SSSS SSSD ; S=SIGNBIT
        LAC  RES1
        SUB  ONE
        BGZ  SATOPH        QUANTIZED SUBBAND SAMPLE GREATER THAN MAX. VALUE?
        ZAC
        SUB  ONE,1
        SUB  RES1
        BGZ  SATOPL        QUANTIZED SUBBAND SAMPLE LESS THAN MIN. VALUE?
        LAC  RES1,14       SAVE QUANTIZED SUBBAND SAMPLE WITHOUT SATURATION
```

```
        SACL RES1          RES1=SD00 0000 0000 0000
        LAC  RES0,14
        SACH RES0          RES0=--DD DDDD DDDD DDDD
        LAC  RES0
        AND  V3FFF         ACCU LOW=00DD DDDD DDDD DDDD
        ADD  RES1          ACCU LOW=SDDD DDDD DDDD DDDD (Q14,Q15,Q18 FORMAT)
        SACL OUTPCM
        RET
SATOPH  LAC  V3FFF         SATURATE QUANTIZED SUBBAND SAMPLE TO MAX. AND SAVE
        ADD  ONE,14
        SACL OUTPCM
        RET
SATOPL  LAC  ONE,15        SATURATE QUANTIZED SUBBAND SAMPLE TO MIN. AND SAVE
        SACL OUTPCM
        RET
*
*                            PCMAQB DECODING DONE
*==============================================================================
```

APPENDIX H

Subroutine INVQMF

```
****************************************************************************
*                           INVQMF FILTERING                               *
*                                                                          *
* THIS SUBROUTINE INCORPORATES:                                            *
*                          -INVERSE QMF FILTER TREE                        *
*                          -8 kHz SPEECH SIGNAL RECONSTRUCTION             *
*                          -TAKE OVER OF NEW VOICING STRATEGY FOR          *
*                           PCM DECODING                                   *
*                                                                          *
* INPUT:  1 kHz SAMPLES (8 STREAMS)                                        *
* OUTPUT: 8 kHz SAMPLES (1 STREAM)                                         *
*--------------------------------------------------------------------------*
* SUBBAND PROCESSING : 1,8,4,5,2,7,3,6                                     *
*                                                                          *
* BIT ALLOCATION :                                                         *
*                ________________1___2___3___4___5___6___7___8____         *
*                VOICED          4   4   3   2   2   0   0   0             *
*                INTERMEDIATE    4   3   2   2   2   2   0   0             *
*                UNVOICED        2   3   3   3   2   2   0   0             *
* FAW's :                                                                  *
*        VOICED >76      INTERMEDIATE>96        UNVOICED>CC                *
*--------------------------------------------------------------------------*
* AUTHOR: ROLAND KLEUTERS                                                  *
* DATE: 29-5-1987                                                          *
****************************************************************************
*
*---------------------------------------------------------------------------
* MACRO'S
*---------------------------------------------------------------------------
*
*    MACRO TO REALIZE FILTERING IN LOW BRANCH OF FIRST INVQMF STAGE
FLTFLO  $MACRO
        ZAC
        LT   XF16
        MPY  CF1
        LTD  XF15
        MPY  CF3
        LTD  XF14
        MPY  CF5
        LTD  XF13
        MPY  CF7
        LTD  XF12
        MPY  CF9
        LTD  XF11
        MPY  CF11
        LTD  XF10
        MPY  CF13
        LTD  XF9
        MPY  CF15
        LTD  XF8
        MPY  CF14
        LTD  XF7
        MPY  CF12
        LTD  XF6
        MPY  CF10
        LTD  XF5
        MPY  CF8
        LTD  XF4
```

```
        MPY  CF6
        LTD  XF3
        MPY  CF4
        LTD  XF2
        MPY  CF2
        LTA  INPF
        MPY  CF0
        APAC                FILTER RESULT HAS TO BE AMPLIFIED WITH 2
        SACL RES0
        SACH RES1
        ADD  RES0
        ADDH RES1
        SACH OUTP           8 kHz OUTPUT OF TMS32010
        LAC  INPF
        SACL XF2
        $END
*
*    MACRO TO REALIZE FILTERING IN HIGH BRANCH OF FIRST INVQMF STAGE
FLTFHI  $MACRO
        ZAC
        LT   XF32
        MPY  CF0
        LTD  XF31
        MPY  CF2
        LTD  XF30
        MPY  CF4
        LTD  XF29
        MPY  CF6
        LTD  XF28
        MPY  CF8
        LTD  XF27
        MPY  CF10
        LTD  XF26
        MPY  CF12
        LTD  XF25
        MPY  CF14
        LTD  XF24
        MPY  CF15
        LTD  XF23
        MPY  CF13
        LTD  XF22
        MPY  CF11
        LTD  XF21
        MPY  CF9
        LTD  XF20
        MPY  CF7
        LTD  XF19
        MPY  CF5
        LTD  XF18
        MPY  CF3
        LTA  INPF
        MPY  CF1
        APAC                FILTER RESULT HAS TO BE AMPLIFIED WITH 2
        SACL RES0
        SACH RES1
        ADD  RES0
        ADDH RES1
        SACH OUTP           8 kHz OUTPUT OF TMS32010
```

```
        LAC   INPF
        SACL  XF18
        $END
*
*    MACRO TO REALIZE FILTERING IN LOW BRANCH OF SECOND INVQMF STAGE
FLTSLO  $MACRO
        ZAC
        LT    XS8
        MPY   CS1
        LTD   XS7
        MPY   CS3
        LTD   XS6
        MPY   CS5
        LTD   XS5
        MPY   CS7
        LTD   XS4
        MPY   CS6
        LTD   XS3
        MPY   CS4
        LTD   XS2
        MPY   CS2
        LTD   INPS
        MPY   CS0
        APAC                  FILTER RESULT HAS TO BE AMPLIFIED WITH 2
        SACL  RES0
        SACH  RES1
        ADD   RES0
        ADDH  RES1
        SACH  RES0            4 kHz INPUT TO FIRST INVQMF STAGE
        $END
*
*    MACRO TO REALIZE FILTERING IN HIGH BRANCH OF SECOND INVQMF STAGE
FLTSHI  $MACRO
        ZAC
        LT    XS8
        MPY   CS0
        LTD   XS7
        MPY   CS2
        LTD   XS6
        MPY   CS4
        LTD   XS5
        MPY   CS6
        LTD   XS4
        MPY   CS7
        LTD   XS3
        MPY   CS5
        LTD   XS2
        MPY   CS3
        LTD   INPS
        MPY   CS1
        APAC                  FILTER RESULT HAS TO BE AMPLIFIED WITH 2
        SACL  RES0
        SACH  RES1
        ADD   RES0
        ADDH  RES1
        SACH  RES0            4 kHz INPUT TO FIRST INVQMF STAGE
        $END
*
```

```
*     MACRO TO REALIZE FILTERING IN LOW BRANCH OF THIRD INVQMF STAGE
FLTTLO  $MACRO
        ZAC
        LT   XT6
        MPY  CT1
        LTD  XT5
        MPY  CT3
        LTD  XT4
        MPY  CT5
        LTD  XT3
        MPY  CT4
        LTD  XT2
        MPY  CT2
        LTD  INPT
        MPY  CT0
        APAC                 FILTER RESULT HAS TO BE AMPLIFIED WITH 2
        SACL RES0
        SACH RES1
        ADD  RES0
        ADDH RES1
        SACH RES0            2 kHz INPUT TO SECOND INVQMF STAGE
        $END
*
*     MACRO TO REALIZE FILTERING IN HIGH BRANCH OF THIRD INVQMF STAGE
FLTTHI  $MACRO
        ZAC
        LT   XT6
        MPY  CT0
        LTD  XT5
        MPY  CT2
        LTD  XT4
        MPY  CT4
        LTD  XT3
        MPY  CT5
        LTD  XT2
        MPY  CT3
        LTD  INPT
        MPY  CT1
        APAC                 FILTER RESULT HAS TO BE AMPLIFIED WITH 2
        SACL RES0
        SACH RES1
        ADD  RES0
        ADDH RES1
        SACH RES0            2 kHz INPUT TO SECOND INVQMF STAGE
        $END
*
*     MACRO TO LOAD DELAY VARIABLES OF SECOND INVQMF STAGE
LOADDS  $MACRO X
        LACK :X.S:
        LARK 0,XS2
        TBLR *+
        ADD  ONE
        TBLR *+
        ADD  ONE
        TBLR *+
        ADD  ONE
        TBLR *+
        ADD  ONE
```

```
        TBLR *+
        ADD  ONE
        TBLR *+
        ADD  ONE
        TBLR *
        $END
*
*     MACRO TO SAVE DELAY VARIABLES OF SECOND INVQMF STAGE
SAVEDS  $MACRO X
        LACK :X.S:
        LARK 0,XS2
        TBLW *+
        ADD  ONE
        TBLW *+
        ADD  ONE
        TBLW *+
        ADD  ONE
        TBLW *+
        ADD  ONE
        TBLW *+
        ADD  ONE
        TBLW *+
        ADD  ONE
        TBLW *
        $END
*
*     MACRO TO LOAD DELAY VARIABLES OF THIRD INVQMF STAGE
LOADDT  $MACRO X
        LACK :X.S:
        LARK 0,XT2
        TBLR *+
        ADD  ONE
        TBLR *+
        ADD  ONE
        TBLR *+
        ADD  ONE
        TBLR *+
        ADD  ONE
        TBLR *
        $END
*
*     MACRO TO SAVE DELAY VARIABLES OF THIRD INVQMF STAGE
SAVEDT  $MACRO X
        LACK :X.S:
        LARK 0,XT2
        TBLW *+
        ADD  ONE
        TBLW *+
        ADD  ONE
        TBLW *+
        ADD  ONE
        TBLW *+
        ADD  ONE
        TBLW *
        $END
*
```

```
*---------------------------------------------------------------------------
* MACRO DEFINITION DONE; MAIN PART OF SUBROUTINE BEGINS
*---------------------------------------------------------------------------
*
INVQMF  NOP                 ENTRY POINT
*
*     SELECT PATH
        LACK 4
        SUB  PATH
        BGZ  PATHGZ
        BZ   HIGH4          SUBBAND 2
        ADD  ONE
        BZ   HIGH6          SUBBAND 7
        ADD  ONE
        BZ   HIGH5          SUBBAND 3
        B    HIGH7          SUBBAND 6
PATHGZ  SUB  ONE
        BZ   LOW7           SUBBAND 5
        SUB  ONE
        BZ   LOW5           SUBBAND 4
        SUB  ONE
        BZ   LOW6           SUBBAND 8
*                           SUBBAND 1
*
*---------------------------------------------------------------------------
* THIRD INVQMF STAGE:
*                   LOW4  0-500     Hz
*                   HIGH4 500-1000  Hz
*                   LOW5  1500-2000 Hz
*                   HIGH5 1000-1500 Hz
*                   LOW6  3500-4000 Hz; NOT CODED
*                   HIGH6 3000-3500 Hz; NOT CODED
*                   LOW7  2000-2500 Hz
*                   HIGH7 2500-3000 Hz
*                   TAKE OVER NEW VOICING STRATEGY
*---------------------------------------------------------------------------
*
LOW4    DMOV A4             DOUBLE BUFFER
*
*     INPUT CALCULATION
        ZALH B4
        SUBH C4
        SACH INPT           1 kHz INPUT TO FILTER
        LAC  OUTPCM
        SACL A4             BUFFERED 1 kHz INPUT TO THIRD INVQMF STAGE
*
*     LOAD DELAY VARIABLES OF LOW4
        LOADDT XXT2
*
*     FILTER 0-500 Hz
        FLTTLO
*
*     SAVE DELAY VARIABLES OF LOW4
        SAVEDT XXT2
*
*     BRANCH TO SECOND INVQMF STAGE
        B    LOW2
*
```

```
HIGH4   NOP                 NO DOUBLE BUFFER
*
*     INPUT CALCULATION
        ZALH C4
        ADDH B4
        SACH INPT           1 kHz INPUT TO FILTER
        LAC  OUTPCM
        SACL C4             BUFFERED 1 kHz INPUT TO THIRD INVQMF STAGE
*
*     LOAD DELAY VARIABLES OF HIGH4
        LOADDT XXT8
*
*     FILTER 500-1000 Hz
        FLTTHI
*
*     SAVE DELAY VARIABLES OF HIGH4
        SAVEDT XXT8
*
*     BRANCH TO SECOND INVQMF STAGE
        B    LOW2
*
LOW5    DMOV A5             DOUBLE BUFFER
*
*     INPUT CALCULATION
        LAC  B5,15
        SUB  C5,15
        SACH INPT           1 kHz INPUT TO FILTER
        LAC  OUTPCM
        SACL A5             BUFFERED 1 kHz INPUT TO THIRD INVQMF STAGE
*
*     LOAD DELAY VARIABLES OF LOW5
        LOADDT XXT14
*
*     FILTER 1500-2000 Hz
        FLTTLO
*
*     SAVE DELAY VARIABLES OF LOW5
        SAVEDT XXT14
*
*     BRANCH TO SECOND INVQMF STAGE
*       B    HIGH2
*
HIGH5   NOP                 NO DOUBLE BUFFER
*
*     INPUT CALCULATION
        LAC  C5,15
        ADD  B5,15
        SACH INPT           1 kHz INPUT TO FILTER
        LAC  OUTPCM
        SACL C5             BUFFERED 1 kHz INPUT TO THIRD INVQMF STAGE
*
*     LOAD DELAY VARIABLES OF HIGH5
        LOADDT XXT20
*
*     FILTER 1000-1500 Hz
        FLTTHI
*
```

```
*     SAVE DELAY VARIABLES OF HIGH5
        SAVEDT XXT20
*
*     BRANCH TO SECOND INVQMF STAGE
        B    HIGH2
*
LOW6    NOP
*
*     THE 3500-4000 Hz BAND IS NOT CODED
*
*     ZERO 2 kHz INPUT TO SECOND INVQMF STAGE
        ZAC
        SACL RES0           2 kHz INPUT TO SECOND INVQMF STAGE
*
*     BRANCH TO SECOND INVQMF STAGE
        B    LOW3
*
HIGH6   NOP
*
*     THE 3000-3500 Hz BAND IS NOT CODED
*
*     ZERO 2 kHZ INPUT TO SECOND INVQMF STAGE
        ZAC
        SACL RES0
*
*     BRANCH TO SECOND INVQMF STAGE
        B    LOW3
*
LOW7    DMOV A7             DOUBLE BUFFER
*
*     INPUT CALCULATION
        LAC  B7,15
        SUB  C7,15
        SACH INPT           1 kHz INPUT TO FILTER
        LAC  OUTPCM
        SACL A7             BUFFERED 1 kHz INPUT TO THIRD INVQMF STAGE
*
*     LOAD DELAY VARIABLES OF LOW7
        LOADDT XXT26
*
*     FILTER 2000-2500 Hz
        FLTTLO
*
*     SAVE DELAY VARIABLES OF LOW7
        SAVEDT XXT26
*
*     BRANCH TO SECOND INVQMF STAGE
        B    HIGH3
*
HIGH7   NOP                 NO DOUBLE BUFFER
*
*     INPUT CALCULATION
        LAC  C7,15
        ADD  B7,15
        SACH INPT           1 kHz INPUT TO FILTER
        LAC  OUTPCM
        SACL C7             BUFFERED 1 kHz INPUT TO THIRD INVQMF STAGE
*
```

```
*     LOAD DELAY VARIABLES OF HIGH7
       LOADDT XXT32
*
*     FILTER 2500-3000 Hz
       FLTTHI
*
*     SAVE DELAY VARIABLES OF HIGH7
       SAVEDT XXT32
*
*     UPDATE FRAME AND TAKE OVER NEW VOICING STRATEGY IF END OF FRAME
       LAR  0,FRAME
       BANZ SFRAME
       LACK 7              END OF FRAME; PREPARE NEXT FRAME
       SACL FRAME
       DMOV STRAT          TAKE OVER NEW VOICING STRATEGY
       B    BRSEC
SFRAME SAR  0,FRAME        NOT END OF FRAME; PREPARE NEXT DATABLOCK
*
*     BRANCH TO SECOND INVQMF STAGE
BRSEC  B    HIGH3
*
*-----------------------------------------------------------------------------
* SECOND INVQMF STAGE:
*                    LOW2  0kHz-1kHz
*                    HIGH2 1kHz-2kHz
*                    LOW3  3kHz-4kHz
*                    HIGH3 2kHz-3kHz
*-----------------------------------------------------------------------------
*
LOW2   DMOV A2             DOUBLE BUFFER
*
*     INPUT CALCULATION
       ZALH B2
       SUBH C2
       SACH INPS           2 kHz INPUT TO FILTER
       LAC  RESO
       SACL A2             BUFFERED 2 kHz INPUT TO SECOND INVQMF STAGE
*
*     LOAD DELAY VARIABLES OF LOW2
       LOADDS XXS2
*
*     FILTER 0-1 kHz
       FLTSLO
*
*     SAVE DELAY VARIABLES OF LOW2
       SAVEDS XXS2
*
*     BRANCH TO FIRST INVQMF STAGE
       B    LOW1
*
HIGH2  NOP                 NO DOUBLE BUFFER
*
*     INPUT CALCULATION
       ZALH C2
       ADDH B2
       SACH INPS           2 kHz INPUT TO FILTER
       LAC  RESO
       SACL C2             BUFFERED 2 kHz INPUT TO SECOND INVQMF STAGE
```

```
*
*     LOAD DELAY VARIABLES OF HIGH2
       LOADDS XXS10
*
*     FILTER 1-2 kHz
       FLTSHI
*
*     SAVE DELAY VARIABLES OF HIGH2
       SAVEDS XXS10
*
*     BRANCH TO FIRST INVQMF STAGE
       B   LOW1
*
LOW3    DMOV A3              DOUBLE BUFFER
*
*     INPUT CALCULATION
       LAC  B3,15
       SUB  C3,15
       SACH INPS             2 kHz INPUT TO FILTER
       LAC  RESO
       SACL A3               BUFFERED 2 kHz INPUT TO SECOND INVQMF STAGE
*
*     LOAD DELAY VARIABLES OF LOW3
       LOADDS XXS18
*
*     FILTER 3-4 kHz
       FLTSLO
*
*     SAVE DELAY VARIABLES OF LOW3
       SAVEDS XXS18
*
*     BRANCH TO FIRST INVQMF STAGE
       B    HIGH1
*
HIGH3   NOP                  NO DOUBLE BUFFER
*
*     INPUT CALCULATION
       LAC  C3,15
       ADD  B3,15
       SACH INPS             2 kHz INPUT TO FILTER
       LAC  RESO
       SACL C3               BUFFERED 2 kHz INPUT TO SECOND INVQMF STAGE
*
*     LOAD DELAY VARIABLES OF HIGH3
       LOADDS XXS26
*
*     FILTER 2-3 kHz
       FLTSHI
*
*     SAVE DELAY VARIABLES OF HIGH3
       SAVEDS XXS26
*
*     BRANCH TO FIRST INVQMF STAGE
       B    HIGH1
*
```

```
*-----------------------------------------------------------------------------
* FIRST INVQMF STAGE:
*                    LOW1  0kHz-2kHz
*                    HIGH1 2kHz-4kHz
*-----------------------------------------------------------------------------
*
LOW1    DMOV A1              DOUBLE BUFFER
*
*     INPUT CALCULATION
        ZALH B1
        SUB  C1,14
        SACH INPF,1          4 kHz INPUT TO FILTER
        LAC  RES0
        SACL A1              BUFFERED 4 kHz INPUT TO FIRST INVQMF STAGE
*
*     FILTER 0-2 kHz
        FLTFLO
*
*     INVQMF DONE
        RET
*
HIGH1   NOP                  NO DOUBLE BUFFER
*
*     INPUT CALCULATION
        LAC  C1,14
        ADDH B1
        SACH INPF,1          4 kHz INPUT TO FILTER
        LAC  RES0
        SACL C1              BUFFERED 4 kHz INPUT TO FIRST INVQMF STAGE
*
*     FILTER 2-4 kHz
        FLTFHI
*
*     INVQMF DONE
        RET
*
*                               INVQMF FILTERING DONE
*=============================================================================
```

APPENDIX I

Subroutine DEMULTIPLEXING

```
***************************************************************************
*                              DEMULTIPLEXING                             *
*                                                                         *
* THIS SUBROUTINE INCORPORATES:                                           *
*                           -RECEIVING OF PCM CODED DATA AND              *
*                           -DEMULTIPLEXING                               *
*                                                                         *
* INPUT:  16 KBPS BITSTREAM                                               *
* OUTPUT: FAW + 1 kHz SAMPLES (8 STREAMS)                                  *
*-------------------------------------------------------------------------*
* SUBBAND PROCESSING : 1,8,4,5,2,7,3,6                                    *
*                                                                         *
* BITALLOCATION :                                                         *
*              ________________1__2__3__4__5__6__7__8___                  *
*              VOICED          4  4  3  2  2  0  0  0                     *
*              INTERMEDIATE    4  3  2  2  2  2  0  0                     *
*              UNVOICED        2  3  3  3  2  2  0  0                     *
* FAW's :                                                                 *
*         VOICED >76        INTERMEDIATE >96         UNVOICED >CC         *
*-------------------------------------------------------------------------*
* AUTHOR: ROLAND KLEUTERS                                                 *
* DATA: 29-5-1987                                                         *
***************************************************************************
*
*--------------------------------------------------------------------------
* MACRO'S
*--------------------------------------------------------------------------
*
*    MACRO TO RECEIVE SIGNBIT AND EXTEND SIGN IN RECEIVE BUFFER
RECSB   $MACRO
        IN   SIGIN,3        SIGNBIT RECEIVED VIA PIN 12 OF EXPANSION PORT
        LAC  ONE
        AND  SIGIN          SELECT SIGNBIT; SIGNBIT IS LSB OF ACCU
        SACL DATA           DATA=>0000 or >0001
        ZAC                 EXTEND SIGN IN RECEIVE BUFFER
        SUB  DATA
        SACL DATA           DATA=>0000 or >FFFF
        $END
*
*    MACRO TO RECEIVE ONE BIT
RCEIVE  $MACRO
        LAC  DATA,1
        SACL DATA           PLACE RESERVED FOR NEW RECEIVE BIT
        IN   SIGIN,3        RECEIVE BIT RECEIVED VIA PIN 12 OF EXPANSION PORT
        LAC  ONE
        AND  SIGIN          SELECT RECEIVE BIT; RECEIVE BIT IS LSB OF ACCU
        XOR  DATA
        SACL DATA           RECEIVE BIT INSERTED INTO LSB OF DATA
        $END
*
*--------------------------------------------------------------------------
* MACRO DEFINITION DONE; MAIN PART OF SUBROUTINE BEGINS
*--------------------------------------------------------------------------
*
DEMUX   NOP                 ENTRY POINT
*
```

```
*     CHECK IF DEMULTIPLEXER IS IN ALLIGNMENT (SWITCH NEQ >FF)
       LACK >FF
       XOR  SWITCH
       BNZ  RCMODE
*
*:::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
*                    DEMULTIPLEXER IS NOT IN ALLIGNMENT
*:::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
*
*     FAW SEARCH OR 128 BIT (=FRAME) SKIP?
       LAC  POINT
       BZ   SKPBIT
*
*     FAW SEARCH; INSERT ONE BIT IN SEARCH FOR A FAW
       RCEIVE
       LACK >FF
       AND  DATA            SELECT 8 LSB's OF DATA
       SACL DATA            DATA CONTAINS 0000 0000 xxxx xxxx
*
*     CHECK FAW
       LACK >76
       XOR  DATA            VOICED?
       BZ   VOIFND
       LACK >96
       XOR  DATA            INTERMEDIATE?
       BZ   INTFND
       LACK >CC
       XOR  DATA            UNVOICED?
       BZ   UNVFND
       RET                  AT NEXT ENTRY AGAIN FAW SEARCH
*
*     FAW FOUND, SET STRATEGY
VOIFND ZAC                  STRATEGY=0 (VOICED)
       SACL STRAT
       B    INSBIT
INTFND LACK 1               STRATEGY=1 (INTERMEDIATE)
       SACL STRAT
       B    INSBIT
UNVFND LACK 2               STRATEGY=2 (UNVOICED)
       SACL STRAT
*
*     INITIALIZATION FOR 128 BIT (=FRAME) SKIP
INSBIT LACK 127
       SACL COUNT           COUNT 128 BITS
       ZAC
       SACL POINT           NEXT ENTRY AT SKPBIT
       RET
*
*     SKIP 128 BITS BEFORE SECOND FAW CHECK
SKPBIT RCEIVE
       LAR  0,COUNT
       BANZ CONTSK
*
```

```
*     128 BITS SKIPPED
       ZAC
       SACL ERROR          NO ERROR ALLOWANCES
       LACK >FF
       AND  DATA           SELECT 8 LSB's OF DATA
       SACL DATA           DATA CONTAINS 0000 0000 xxxx xxxx
       B    FAWCHK
*
*     128 BITS NOT YET SKIPPED
CONTSK SAR  0,COUNT
       RET                 NEXT ENTRY AGAIN AT SKPBIT
*
*---------------------------------------------------------------------------
* Frame Allignment Word  CONFIRMATION
*---------------------------------------------------------------------------
*
FAWCHK NOP
       LACK >76
       XOR  DATA           VOICED?
       BZ   CONF0
       LACK >96
       XOR  DATA           INTERMEDIATE?
       BZ   CONF1
       LACK >CC
       XOR  DATA           UNVOICED?
       BZ   CONF2
       LAR  0,ERROR        FAW NOT CONFIRMED; ERROR ALLOWED?
       BANZ ERR            ERROR ALLOWED?
*
*     ERROR NOT ALLOWED; SO SWITCH TO MISALLIGNMENT
       LACK >FF
       SACL SWITCH         TURN OFF DECODER
       SACL POINT
       RET                 NEXT ENTRY AT FRAME SEARCH
*
*     ERROR ALLOWED; DECREMENT NO. OF ERROR ALLOWANCES; STRATEGY UNCHANGED
ERR    SAR  0,ERROR
       B    INISYN
*
*     FAW CONFIRMED; SET STRATEGY AND NO. OF ERROR ALLOWANCES
CONF0  ZAC
       SACL STRAT          SET VOICED STRATEGY (=0)
       LACK 4
       SACL ERROR          4 FRAMES IN ERROR ALLOWANCE
       B    INISYN
CONF1  LACK 1
       SACL STRAT          SET INTERMEDIATE STRATEGY (=1)
       LACK 4
       SACL ERROR          4 FRAMES IN ERROR ALLOWANCE
       B    INISYN
CONF2  LACK 2
       SACL STRAT          SET UNVOICED STRATEGY (=2)
       LACK 4
       SACL ERROR          4 FRAMES IN ERROR ALLOWANCE
*
```

```
*     INITIALIZE FOR A NEW FRAME
INISYN  LACK >FF
        SACL DEMO           NEW ACCESS MARK
        LACK 1
        SACL BAND           NEXT SUBBAND TO RECEIVE (0-500 Hz)
*
*     SYNCHRONIZE DECODER IF NEEDED
        LAC  SWITCH
        BZ   INIOUT         DECODER ALREADY ON, i.e. SYNCHRONIZATION NEEDED?
        LACK 4
        SACL SWITCH         DELAY FOR DECODER TO SWITCH ON
        LACK 13
        SACL STATE          SET STATE TO 13 (STATE=0 to 15)
        LACK 7
        SACL FRAME          INITIALIZE DECODER STATUS
INIOUT  RET                 NEXT ENTRY AT RECEIVE MODE
*
*::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
*              DEMULTIPLEXER IS IN ALLIGNMENT; RECEIVE MODE
*::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
*
RCMODE  NOP
*
*     CHECK IF DECODER IS ON (SWITCH=0); IF NOT DECREMENT SWITCH
        ZALS SWITCH
        BZ   FAWSUB
        SUB  ONE
        SACL SWITCH
*
*     SELECT FAW or ONE OF THE SUBBANDS FROM WHICH DATA HAS TO BE SENT
FAWSUB  LACK 4
        SUB  BAND
        BGZ  FWSBGZ
        BZ   NEXT4          SUBBAND 4 (1500-2000 Hz)
        ADD  ONE
        BZ   NEXT5          SUBBAND 5 (2000-2500 Hz)
        B    NEXT6          SUBBAND 6 (2500-3000 Hz)
FWSBGZ  SUB  ONE
        BZ   NEXT3          SUBBAND 3 (1000-1500 Hz)
        SUB  ONE
        BZ   NEXT2          SUBBAND 2 (500-1000 Hz)
        SUB  ONE
        BZ   NEXT1          SUBBAND 1 (0-500 Hz)
*                           FAW
*
*-----------------------------------------------------------------------------
* RECEIVE FAW BITS
*-----------------------------------------------------------------------------
*
NFAW    ZALS DEMO           NEW ACCESS CHECK
        BZ   FSIG
*
*     INITIALIZATION FOR A NEW FAW ACCESS
        ZAC
        SACL DEMO           NO NEW ACCESS ANYMORE
        SACL DATA           CLEAR RECEIVE BUFFER
        LACK 7
        SACL COUNT          8 FAW BITS TO RECEIVE
```

```
*
*     RECEIVE A FAW BIT AND INSERT IT INTO LSB OF DATA
FSIG    RCEIVE
        LAR  0,COUNT
        BANZ CONT
*
*     ALL FAW BITS RECEIVED; CHECK FAW
        B    FAWCHK
*
*     NOT ALL FAW BITS RECEIVED,PREPARE NEXT BIT
CONT    SAR  0,COUNT
        RET
*
*-----------------------------------------------------------------------------
* RECEIVE SUBBAND BITS
*-----------------------------------------------------------------------------
*
*.............................................................................
*               A BIT OF SUBBAND 1 (0-500 Hz) HAS TO BE RECEIVED
*.............................................................................
*
NEXT1   ZALS DEMO           NEW ACCESS CHECK
        BZ   LOOP1
*
*     INITIALIZATION FOR A NEW SUBBAND 1 ACCESS
        ZAC
        SACL DEMO           NO NEW ACCESS ANYMORE
        LACK 2
        XOR  STRAT          UNVOICED?
        BNZ  VOIN1
*
*     FURTHER INITIALIZATION FOR UNVOICED STRATEGY
        ZAC
        SACL COUNT          1 DATABIT TO RECEIVE
        B    SIGN1
*
*     FURTHER INITIALIZATION FOR VOICED AND INTERMEDIATE STRATEGY
VOIN1   LACK 2
        SACL COUNT          3 DATABITS TO RECEIVE
*
*     RECEIVE THE SIGNBIT OF SUBBAND 1 AND EXTEND THE SIGN IN RECEIVE BUFFER
SIGN1   RECSB
        RET
*
*     RECEIVE A SUBBAND 1 DATABIT AND INSERT IT INTO LSB OF DATA
LOOP1   RCEIVE
        LAR  0,COUNT
        BANZ CONT1
*
*     ALL SUBBAND 1 BITS RECEIVED; UPDATE COD1 AND PREPARE NEXT SUBBAND
        LACK COD1
        TBLW DATA
        LACK >FF
        SACL DEMO           NEW ACCESS MARK
        LACK 4
        SACL BAND           NEXT SUBBAND TO RECEIVE (1500-2000 Hz)
        RET
*
```

```
*     NOT ALL SUBBAND 1 BITS RECEIVED,PREPARE NEXT BIT
CONT1   SAR  0,COUNT
        RET
*
*.............................................................................
*              A BIT OF SUBBAND 2 (500-1000 Hz) HAS TO BE RECEIVED
*.............................................................................
*
NEXT2   ZALS DEMO           NEW ACCESS CHECK
        BZ   LOOP2
*
*     INITIALIZATION FOR A NEW SUBBAND 2 ACCESS
        ZAC
        SACL DEMO           NO NEW ACCESS ANYMORE
        LAC  STRAT          VOICED?
        BNZ  INUN2
*
*     FURTHER INITIALIZATION FOR VOICED STRATEGY
        LACK 2
        SACL COUNT          3 DATABITS TO RECEIVE
        B    SIGN2
*
*     FURTHER INITIALIZATION FOR INTERMEDIATE AND UNVOICED STRATEGY
INUN2   LACK 1
        SACL COUNT          2 DATABITS TO RECEIVE
*
*     RECEIVE THE SIGNBIT OF SUBBAND 2 AND EXTEND THE SIGN IN RECEIVE BUFFER
SIGN2   RECSB
        RET
*
*     RECEIVE A SUBBAND 2 DATABIT AND INSERT IT INTO LSB OF DATA
LOOP2   RCEIVE
        LAR  0,COUNT
        BANZ CONT2
*
*     ALL SUBBAND 2 BITS RECEIVED; UPDATE COD2 AND PREPARE NEXT SUBBAND
        LACK COD2
        TBLW DATA
        LACK >FF
        SACL DEMO           NEW ACCESS MARK
        LACK 3
        SACL BAND           NEXT SUBBAND TO RECEIVE (1000-1500 Hz)
        RET
*
*     NOT ALL SUBBAND 2 BITS RECEIVED,PREPARE NEXT BIT
CONT2   SAR  0,COUNT
        RET
*
*.............................................................................
*            A BIT OF SUBBAND 3 (1000-1500 Hz) HAS TO BE RECEIVED
*.............................................................................
*
NEXT3   ZALS DEMO           NEW ACCESS CHECK
        BZ   LOOP3
*
```

```
*     INITIALIZATION FOR A NEW SUBBAND 3 ACCESS
        ZAC
        SACL DEMO           NO NEW ACCESS ANYMORE
        LACK 1
        XOR  STRAT          INTERMEDIATE?
        BNZ  VOUN3
*
*     FURTHER INITIALIZATION FOR INTERMEDIATE STRATEGY
        ZAC
        SACL COUNT          1 DATABIT TO RECEIVE
        B    SIGN3
*
*     FURTHER INITIALIZATION FOR VOICED AND UNVOICED STRATEGY
VOUN3   LACK 1
        SACL COUNT          2 DATABITS TO RECEIVE
*
*     RECEIVE THE SIGNBIT OF SUBBAND 3 AND EXTEND THE SIGN IN RECEIVE BUFFER
SIGN3   RECSB
        RET
*
*     RECEIVE A SUBBAND 3 DATABIT AND INSERT IT INTO LSB OF DATA
LOOP3   RCEIVE
        LAR  0,COUNT
        BANZ CONT3
*
*     ALL SUBBAND 3 BITS RECEIVED AND PERHAPS ALSO END OF 15 BIT DATABLOCK
*     UPDATE COD3 AND PREPARE NEXT THING TO RECEIVE
        LACK COD3
        TBLW DATA
        LACK >FF
        SACL DEMO           NEW ACCESS MARK
        LAC  STRAT          VOICED?
        BZ   NEXT6F         END OF 15 BIT DATABLOCK
        LACK 6
        SACL BAND           NEXT SUBBAND TO RECEIVE (2500-3000 Hz)
        RET
*
*     NOT ALL SUBBAND 3 BITS RECEIVED,PREPARE NEXT BIT
CONT3   SAR  0,COUNT
        RET
*
*...............................................................................
*           A BIT OF SUBBAND 4 (1500-2000 Hz) HAS TO BE RECEIVED
*...............................................................................
*
NEXT4   ZALS DEMO           NEW ACCESS CHECK
        BZ   LOOP4
*
*     INITIALIZATION FOR A NEW SUBBAND 4 ACCESS
        ZAC
        SACL DEMO           NO NEW ACCESS ANYMORE
        LACK 2
        XOR  STRAT          UNVOICED?
        BNZ  VOIN4
*
```

```
*     FURTHER INITIALIZATION FOR UNVOICED STRATEGY
        LACK 1
        SACL COUNT          2 DATABITS TO RECEIVE
        B    SIGN4
*
*     FURTHER INITIALIZATION FOR VOICED AND INTERMEDIATE STRATEGY
VOIN4   ZAC
        SACL COUNT          1 DATABIT TO RECEIVE
*
*     RECEIVE THE SIGNBIT OF SUBBAND 4 AND EXTEND THE SIGN IN RECEIVE BUFFER
SIGN4   RECSB
        RET
*
*     RECEIVE A SUBBAND 4 DATABIT AND INSERT IT INTO LSB OF DATA
LOOP4   RCEIVE
        LAR  0,COUNT
        BANZ CONT4
*
*     ALL SUBBAND 4 BITS RECEIVED; UPDATE COD4 AND PREPARE NEXT SUBBAND
        LACK COD4
        TBLW DATA
        LACK >FF
        SACL DEMO           NEW ACCESS MARK
        LACK 5
        SACL BAND           NEXT SUBBAND TO RECEIVE (2000-2500 Hz)
        RET
*
*     NOT ALL SUBBAND 4 BITS RECEIVED,PREPARE NEXT BIT
CONT4   SAR  0,COUNT
        RET
*
*.............................................................................
*          A BIT OF SUBBAND 5 (2000-2500 Hz) HAS TO BE RECEIVED
*.............................................................................
*
NEXT5   ZALS DEMO           NEW ACCESS CHECK
        BZ   LOOP5
*
*     INITIALIZATION FOR A NEW SUBBAND 5 ACCESS
        ZAC
        SACL DEMO           NO NEW ACCESS ANYMORE
*                           1 DATABIT TO RECEIVE FOR ALL STRATEGIES
*
*     RECEIVE THE SIGNBIT OF SUBBAND 5 AND EXTEND THE SIGN IN RECEIVE BUFFER
SIGN5   RECSB
        RET
*
*     RECEIVE THE SUBBAND 5 DATABIT AND INSERT IT INTO LSB OF DATA
LOOP5   RCEIVE
*
*     ALL SUBBAND 5 BITS RECEIVED; UPDATE COD5 AND PREPARE NEXT SUBBAND
        LACK COD5
        TBLW DATA
        LACK >FF
        SACL DEMO           NEW ACCESS MARK
        LACK 2
        SACL BAND           NEXT SUBBAND TO RECEIVE (500-1000 Hz)
        RET
```

```
*
*.........................................................................
*         A BIT OF SUBBAND 6 (2500-3000 Hz) HAS TO BE RECEIVED
*         ONLY ACCESSED  BY UNVOICED AND INTERMEDIATE STRATEGY
*.........................................................................
*
NEXT6   ZALS DEMO           NEW ACCESS CHECK
        BZ   LOOP6
*
*     INITIALIZATION FOR A NEW SUBBAND 6 ACCESS
        ZAC
        SACL DEMO           NO NEW ACCESS ANYMORE
*                           1 DATABIT TO RECEIVE FOR ALL POSSIBLE STRATEGIES
*
*     RECEIVE THE SIGNBIT OF SUBBAND 6 AND EXTEND THE SIGN IN RECEIVE BUFFER
SIGN6   RECSB
        RET
*
*     RECEIVE THE SUBBAND 6 DATABIT AND INSERT IT INTO LSB OF DATA
LOOP6   RCEIVE
*
*     ALL SUBBAND 6 BITS RECEIVED AND ALSO END OF 15 BIT DATABLOCK
*     UPDATE COD6 AND CHECK FOR END OF FRAME
        LACK COD6
        TBLW DATA
        LACK >FF
        SACL DEMO           NEW ACCESS MARK
NEXT6F  LAC  FRAME
        BNZ  NENDFM
*
*     END OF FRAME,PREPARE NEXT FRAME
        ZAC
        SACL BAND           NEXT TO RECEIVE=FAW
        RET
*
*     NOT END OF FRAME,PREPARE NEXT DATABLOCK
NENDFM  LACK 1
        SACL BAND           NEXT SUBBAND TO RECEIVE (0-500 Hz)
        RET
*
*                           DEMULTIPLEXING DONE
*=========================================================================
```