Design and Implementation of the DAB/DMB Transmitter Identification Information Decoder

Hongsheng Zhang, Hongyun Wang, Guoyu Wang* and Mingying Lu

***Abstract*—**The Transmitter Identification Information (TII) provides unambiguous identification of each transmitter in a Digital Audio Broadcast (DAB) and Digital Multimedia Broadcast (DMB) network. Recent researches showed that some useful services, such as location and automatic emergency alert, can be efficiently implemented with the assistance of TII. However many DAB/DMB receivers do not have the TII decoding functionality because the implementation of TII is optional in the standard. This has blocked the application of the new services. In this paper, the TII coding theory is analyzed and the design method of the TII decoder is reported. The proposed method can be either implemented in software, enabling the software-based DAB/DMB receivers to add TII decoding ability simply though firmware updating, or embedded in the hardware of a DAB/DMB baseband chip at a very low cost of only 286 logic elements and 1280 memory bits.

***Keywords*—**Digital Audio Broadcast (DAB), Digital Multimedia Broadcast (DMB), Transmitter Identification Information (TII), TII Decoder.

I. Introduction

DIGITAL Audio Broadcast (DAB) [1], together with its successor, Digital Audio Broadcast Plus (DAB+) [2] have been widely used in many countries as the next generation audio broadcast standard. Besides the high quality audio services, DAB can also broadcast the multimedia information including video, picture, data and text etc. For this reason, DAB is also called Digital Multimedia Broadcast (DMB), which has been standardized in 2005 [3].

DAB, DAB+ and DMB can all be operated in Single Frequency Network (SFN), in which all the transmitters simultaneously broadcast the same signal over the same frequency. The SFN can increase the coverage area, decrease the outage probability, and eliminate the problem of having to retune the radio when traveling from one area to another [4]. In order to unambiguously identify each transmitter in a SFN network, a Transmitter Identity Information (TII) is assigned to each transmitter as its unique ID. When used in conjunction with other service information, the TII feature can provide an estimate of the geographical position of the receiver [1].

The TII is repeatedly transmitted in the first symbol, also called the null symbol, of the DAB, DAB+ and DMB transmission frame. Because the structure of the transmission frame and the TII modulation method of DAB, DAB+ and DMB are exactly the same, only the term ‘DAB’ will be used in the following text, unless otherwise stated.

The implementation of TII is optional in the DAB standard. In most cases, when a DAB receiver travels from one transmitter to another in a SFN, the receiving will not be interrupted because each DAB transmitter is broadcasting the same content. For a common DAB receiver, it does not need to know where the receiving signal comes from, and consequently, not need to decode the TII information.

In recent years, with the popular of DAB, DAB+ and DMB, many researches were carried out to find new possible applications based on these systems [5]-[9]. Some researchers have noticed that new services, such as the location of a DAB receiver and automatic emergency alert with area selection [8], [9], can be implemented with the assistance of TII. However, many DAB receivers on the market do not have the TII decoding functionality and the related researches are few. This paper analyzes the TII coding theory and the implementation method of the TII decoder. Considering there are both software-based and hardware-based DAB demodulators, the method is optimized to make it easily implemented in both software and hardware.

The following sections are organized as follows. In Section II, the TII coding theory is introduced. In Section III, the TII decoding algorithm is analyzed and optimized. In Section IV, two implementation methods of the TII decoder are described. In Section V, the design is tested and the results are analyzed. Finally conclusions are reached in Section VI

II. The Coding Theory of TII

A DAB transmission frame is composed of a null symbol, 1 to 4 Fast Information Channel (FIC) symbols, and several Main Service Channel (MSC) symbols, as illustrated in Fig. 1. During the null symbol, the transmit power is very low as if the transmitter is shut down. This enables the receiver to fast detect a DAB signal and establish coarse time synchronization by just detecting the receiving power [10].

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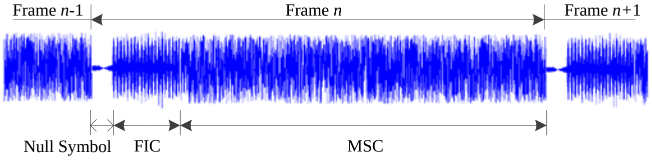

Fig. 1. The structure of a DAB transmission frame

Each DAB symbol is composed by N orthogonal subcarriers in frequency domain. This also applies for the null symbol, which can be seen as composed by N zero-amplitude subcarriers if no TII is added. If a DAB transmitter needs to broadcast its identification in a SFN, the information is modulated only on a limited numbers of subcarriers of the null symbol, so the total energy power of the null symbol is negligibly affected, and the coarse time synchronization algorithm of the DAB receiver needs not to be modified.

The DAB standard defines 4 transmission modes. Each mode has different numbers of subcarriers and symbols. Because most countries use DAB transmission mode I, this paper focuses on the study and implementation of the TII decoder for DAB transmission mode I only. Without loss of generality, the proposed method can also be applied for other transmission modes with minor revisions.

A DAB transmitter identification has 12 bits, composed by a 7-bit Main Identifier, denoted as p with a range of [0, 69], and a 5-bit Sub-Identifier, denoted as c with a range of [0, 23]. The transmitter identification is carried in the null symbol, and the geographical information of each transmitter is carried in the FIC. For any DAB receiver, it has to decode the multiplex configurations information (MCI) in FIC to select a DAB program, so it will know the map of the locations of all the transmitters in a SFN if it also decodes the geographical information in the FIC. Furthermore, if a DAB receiver has the ability to decode the TII, it will know from which transmitter it is receiving, and therefore, the rough location of its position. Based on this theory, the new DAB services, such as geographical location and directional broadcast can be implemented.

For DAB transmission mode I, the 7 bit of p and the 5 bits of c of DAB transmitter identification are modulated on the 1536 subcarriers by the following rules.

Firstly, an amplitude coefficient $A_{c,p}(k)$ is assigned to each subcarrier, where k is the index of the subcarriers, and the values of $A_{c,p}(k)$ are defined by (1).

$$A_{c,p}(k)=\begin{cases} 0 & for \quad k=0,-769 \\ \sum_{b=0}^{7}\delta(k,-768+2c+48b).a_b(p) & for \quad -768\le k<-384 \\ \sum_{b=0}^{7}\delta(k,-384+2c+48b).a_b(p) & for \quad -384\le k<0 \\ \sum_{b=0}^{7}\delta(k,1+2c+48b).a_b(p) & for \quad 0\le k<384 \\ \sum_{b=0}^{7}\delta(k,385+2c+48b).a_b(p) & for \quad 384\le k<768 \end{cases} \tag{1}$$

Secondly, for each subcarrier k, its amplitude is decided to 1 if either $A_{c,p}(k)$ or $A_{c,p}(k\text{-}1)$ is nonzero, or 0 if both $A_{c,p}(k)$ and $A_{c,p}(k\text{-}1)$ are zeros.

The value of $a_b(p)$ in (1) can be found in Table I [1] and δ is the Kronecker symbol defined as

$$\delta(i,j)=\begin{cases} 1 & if \quad i=j \\ 0 & if \quad i\ne j \end{cases} \tag{2}$$

Table I. TII Pattern for DAB Transmission Modes I

p	$a_b(p)$ b=0,1,2…7	p	$a_b(p)$ b=0,1,2…7	p	$a_b(p)$ b=0,1,2…7
0	0 0 0 0 1 1 1	24	0 1 0 1 1 1 0 0	48	1 0 1 0 1 0 0
1	0 0 0 1 0 1 1	25	0 1 1 0 0 0 1 1	49	1 0 1 0 1 0 1
2	0 0 0 1 1 0 1	26	0 1 1 0 0 1 0 1	50	1 0 1 0 1 1 0
3	0 0 0 1 1 1 0	27	0 1 1 0 0 1 1 0	51	1 0 1 1 0 0 0
4	0 0 0 1 1 1 1	28	0 1 1 0 1 0 0 1	52	1 0 1 1 0 0 1
5	0 0 1 0 0 1 1	29	0 1 1 0 1 0 1 0	53	1 0 1 1 0 1 0
6	0 0 1 0 1 0 1	30	0 1 1 0 1 1 0 0	54	1 0 1 1 1 0 0
7	0 0 1 0 1 1 0	31	0 1 1 1 0 0 0 1	55	1 1 0 0 0 0 1
8	0 0 1 0 1 1 1	32	0 1 1 1 0 0 1 0	56	1 1 0 0 0 1 0
9	0 0 1 1 0 0 1	33	0 1 1 1 0 1 0 0	57	1 1 0 0 0 1 1
10	0 0 1 1 0 1 0	34	0 1 1 1 1 0 0 0	58	1 1 0 0 1 0 0
11	0 0 1 1 0 1 1	35	1 0 0 0 0 1 1 1	59	1 1 0 0 1 0 1
12	0 0 1 1 1 0 0	36	1 0 0 0 1 0 1 1	60	1 1 0 0 1 1 0
13	0 0 1 1 1 0 1	37	1 0 0 0 1 1 0 1	61	1 1 0 1 0 0 0
14	0 0 1 1 1 1 0	38	1 0 0 0 1 1 1 0	62	1 1 0 1 0 0 1
15	0 1 0 0 0 1 1	39	1 0 0 1 0 0 1 1	63	1 1 0 1 0 1 0
16	0 1 0 0 1 0 1	40	1 0 0 1 0 1 0 1	64	1 1 0 1 1 0 0
17	0 1 0 0 1 1 0	41	1 0 0 1 0 1 1 0	65	1 1 1 0 0 0 0
18	0 1 0 0 1 1 1	42	1 0 0 1 1 0 0 1	66	1 1 1 0 0 0 1
19	0 1 0 1 0 0 1	43	1 0 0 1 1 0 1 0	67	1 1 1 0 0 1 0
20	0 1 0 1 0 1 0	44	1 0 0 1 1 1 0 0	68	1 1 1 0 1 0 0
21	0 1 0 1 0 1 1	45	1 0 1 0 0 0 1 1	69	1 1 1 1 0 0 0
22	0 1 0 1 1 0 0	46	1 0 1 0 0 1 0 1		
23	0 1 0 1 1 0 1	47	1 0 1 0 0 1 1 0		

To better understand the TII coding theory, it can be summarized by the following rules:

1) The 12 bits of the transmitter identification are actually mapped on 384 subcarriers. For transmission modes with more than 384 subcarriers, the next 384 subcarriers will be repeated and so on. There are 1536 subcarriers in DAB transmission mode I, so the TII will be modulated 4 times.

2) Each 384 subcarriers are divided into 8 sections, each of which has 48 subcarriers. And each section is further divided into 24 subcarrier pairs.

3) The parameter c corresponds to the position of one subcarrier pair in each section. Since there are 8 subcarrier sections, there will be 8 subcarrier pairs selected for every 384 subcarriers, just like a comb.

4) The parameter p defines the pattern of the comb by eliminating half teeth depending on the value of p and $a_b(p)$.

As an example, the amplitudes of all the subcarriers for the TII with c = 2 and p = 3 in DAB transmission mode I is illustrated in Fig. 2.

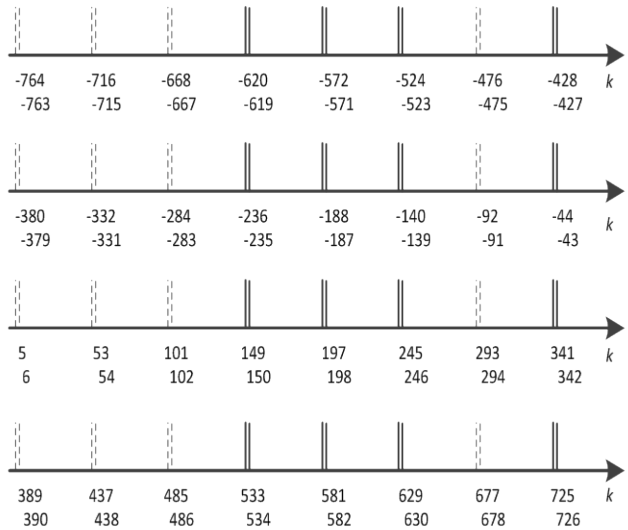

Fig. 2. Example of the subcarriers of the TII (with c = 2 and p = 3) in DAB transmission mode I

III. Design of the TII Decoding Algorithm

In order to decode the TII modulated on the subcarriers of the null symbol, the DAB receiver needs to conduct a time to frequency domain transformation, which is usually done by a Fast Fourier Transformation (FFT), to get the amplitudes of each subcarrier. If an appropriate threshold A_{th} is set, the subcarriers with amplitudes above A_{th} will be considered as modulated subcarriers, and then the values of c and p can be deduced, and consequently, the TII can be decoded.

However, due to the multipath propagation effects in the wireless environment, the receiving signal strength of the DAB receiver is always variant. Although the automatic gain control (AGC) circuit of the DAB receiver tries to keep the receiving signal strength stable, there will be unavoidable fluctuations if the multipath propagation effect is heavy or the DAB receiver moves fast [11]. This makes it difficult to select a fixed A_{th} because the received signal strength is not constant. So a relative and adjustable A_{th} is needed to find those modulated subcarriers. In other words, a subcarrier can be flagged as 'modulated' if its amplitude is considerably bigger than the average.

There is another factor, the Signal Noise Ratio (SNR), which affects the accuracy of the TII decoding algorithm. The SNR of the received signal is also variant. The SNR becomes low if the DAB signal is weak or the noise is big. If the SNR is low to a certain level, the amplitude of the noise can be as high as the same level of the TII subcarriers. In this situation, the selection of the valid subcarrier is difficult and a fake TII may be decoded. So a good TII decoder must also consider the reliability of the algorithm, especially in the low SNR situation. Since the TII is transferred up to 4 times in a DAB transmission frame (as shown in Fig. 2), we can improve the decoding reliability by judging the consistency of 4 decoding results, or even more, if the following DAB transmission frames are taken into consideration.

Considering all the factors above, a TII decoding algorithm is designed as illustrated in Fig. 3.

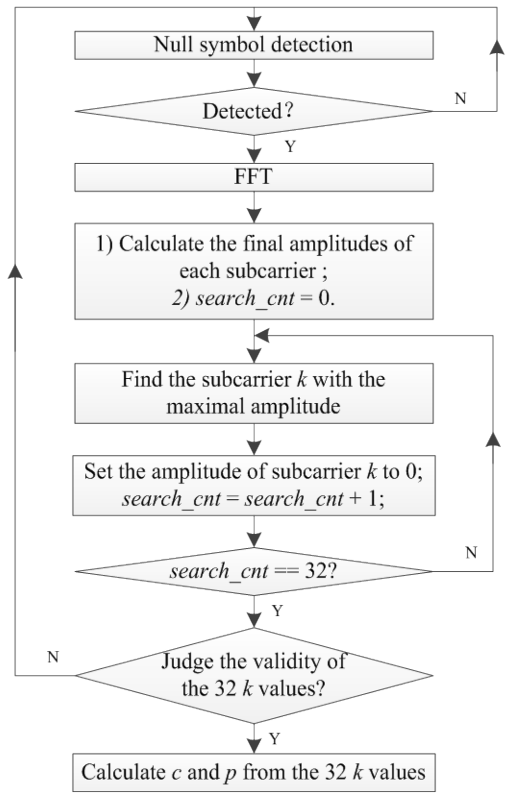

Fig. 3. The flowchart of the proposed TII decoding algorithm

The detailed decoding procedure for DAB transmission mode I is described as follows.

1) Find the null symbol by detecting the falling edge and the rising edge of the DAB receiving signal power. This is a necessary block for all the DAB demodulators.

2) Perform FFT to the null symbol to calculate the real and imaginary amplitudes of each subcarrier (FFT is a complex operation). This step is the same as conducted to other DAB symbols.

3) Calculate the energy of each subcarrier. By definition, the energy of a subcarrier is calculated by the square root of the sum of the squares of its real and imaginary amplitudes. However, this will significantly increase the complexity and computations of the algorithm. As an alternative, we use the average of the absolute values of the real and imaginary amplitudes of each subcarrier as its energy.

4) Search all the subcarriers, record the k value of the subcarrier with the maximal energy, and set its energy to 0 after the search.

5) Repeat Step-4 32 times to find the positions of the 32 maximal-energy subcarriers. Sort the 32 k values in ascending order, and denote the vector as [I_0, I_1, … I_{31}]. The vector is considered valid only if the following conditions are met.

$$\begin{cases} I_{2t+1} - I_{2t} = 1, & \textit{for} \quad t = 0,1,2,\ldots 16 \\ I_{2t} - I_{2t-8} = 384, & \textit{for} \quad t = 4,5,\ldots 16 \end{cases} \tag{3}$$

6) If the detection in Step-5 is valid, then the values of c and $B(p)$ can be decided by the following equations, where $B(p)$ is a

8-bit binary number composed of [$a_0(p)$, $a_1(p)$, …, $a_7(p)$], with $a_0(p)$ being the Most Significant Bit (MSB) and $a_7(p)$ being the Least Significant Bit (LSB).

$$c = \left(I_0 >> 1\right)\%24 \tag{4}$$

$$B(p) = \sum_{t=0}^{3} 2^{7-\lfloor I_{2t}/48 \rfloor} \tag{5}$$

The value of p can then be decided by the value of $B(p)$ through an inverse look-up table (LUT) of Table I. The inverse table has 256 addresses since $B(p)$ has 8 bits. However, $B(p)$ has only 70 meaningful values, so the table contents corresponding to the rest 186 values of $B(p)$ should be set to 127, a false value of p, to avoid misjudgment.

IV. Implementation of the TII Decoder

A. Software Implementation

A software-based DAB receiver means its demodulator performs FFT either by pure software (e.g. the demodulator implemented with powerful Digital Signal Processor), or by an FFT coprocessor whose results can be accessed by software (e.g. the demodulator implemented in Application Specific Integrated Circuit chip with embedded microprocessor).

The TII decoding algorithm proposed above can be easily programmed in software because all the relevant operations are basic operations. If we package the program into a function named as 'tii_decode' which uses a vector of 1536 complex subcarrier amplitudes (i.e. the FFT result) as input, and outputs the values of c, p and the validity flag, then the TII decoder can be simply implemented in a software-based DAB receiver in the following steps:

1) Turn on FFT to the null symbol immediately after it is detected. This step is performed every DAB frame.
2) Read the FFT result from the memory or via the interface of the FFT coprocessor.
3) Invoke the 'tii_decode' function and transfer the FFT result as its input to calculate the values of c and p.

The program and the above steps can be compiled in the form of update package, which enables the DAB receiver to add the TII decoding functionality simply through firmware updating.

B. Hardware Implementation

Some DAB receivers target for low power consumption and low cost, and they use pure hardware demodulators [12], [13]. For these receivers, it may be impracticable to add the TII decoding ability by firmware updating. So a hardware TII decoding method, which can be integrated in the demodulator as hardware Intellectual Property (IP), is also proposed in this paper.

Because the FFT algorithm of the null symbol is same as the other DAB symbols, the FFT circuit can be shared to reduce the hardware consumption. The processing of the FFT results, however, is different between the null symbol and the other symbols. Different from the FIC and MSC symbols, whose FFT results feed to the Differential Quadrature Phase Shift Keying (DQPSK) demodulator for further processing, the null symbol is followed by the TII decoder, where the amplitudes of the FFT results are calculated and processed to extract the information of c and p of the TII. The output of the null symbol detector can be used as a switch signal to decide whether the FFT results flow to the DQPSK demodulator or the TII decoder.

The first 5 steps of the TII decoding algorithm in Section III contain only add, compare and address operations, which can all be easily implemented in hardware. But the algorithm in Step-6 has remainder calculations and divide operations, which must be converted or simplified for hardware implementation.

As for the operation of $(I_0>>1)\%24$, since the max value of I_0 is 240 according to the DAB standard, the result of $(I_0>>1)\%24$ can be calculated by repeating of the compare and subtract operations up to 5 times. Although the required repetition numbers for different I_0 are different, a fixed repetition number of 5 can be used because additional remainder calculations will not affect the final result. This helps to reduce the complexity of the control logic.

As for the calculation of $B(p)$, since the maximal value of I_{2t} is 384, $\lfloor I_{2t}/48 \rfloor$ can then be rewritten as:

$$\lfloor I_{2t}/48 \rfloor = \lfloor I_{2t}[8:0]/48 \rfloor = \lfloor I_{2t}[8:4]/3 \rfloor \tag{6}$$

It can be seen that only the 5 MSBs of I_{2t} are needed to calculate $\lfloor I_{2t}/48 \rfloor$. And because the result of (6) has only 8 possible values, the calculation of $\lfloor I_{2t}/48 \rfloor$ can be easily implemented in hardware using an 8-branch comparator. The result of $B(p)$ can then be calculated by resetting its initial value to 0, and setting the corresponding MSB with the index of $\lfloor I_{2t}/48 \rfloor$ to 1.

A state machine is needed to control each part of the hardware to work orderly. And an address generator is designed to generate the desired buffer address in each stage.

Based on the implementation method stated above, a hardware TII decoder is proposed as in Fig. 4. The final results are stored in an on-chip register, which can be accessed by the extern processor for further processing. The TII decoder is independent from other blocks of the DAB demodulator except the FFT circuit. But since FFT are executed in different time slots, minor modifications are required to the existing FFT interface. Therefore, the proposed hardware TII decoder can be easily integrated in any DAB modulator ASIC chips.

V. Test

Because the relative amplitude detection algorithm is used, the proposed TII decoder is not sensitive to the fluctuation of the received signal strength. However the decoder will fail if the SNR is low to a certain degree. Fig. 5 and Fig. 6 gives the TII decoding result for SNR = 3dB and SNR = 7dB, respectively. In each figure, the upper part is the original TII, the middle part is the subcarriers after FFT, and the lower part is the decoded TII. It can be seen that at the condition of SNR = 3dB, the amplitudes of some TII subcarriers are only slightly bigger than that of the noise, causing the possible failure of a TII detection.

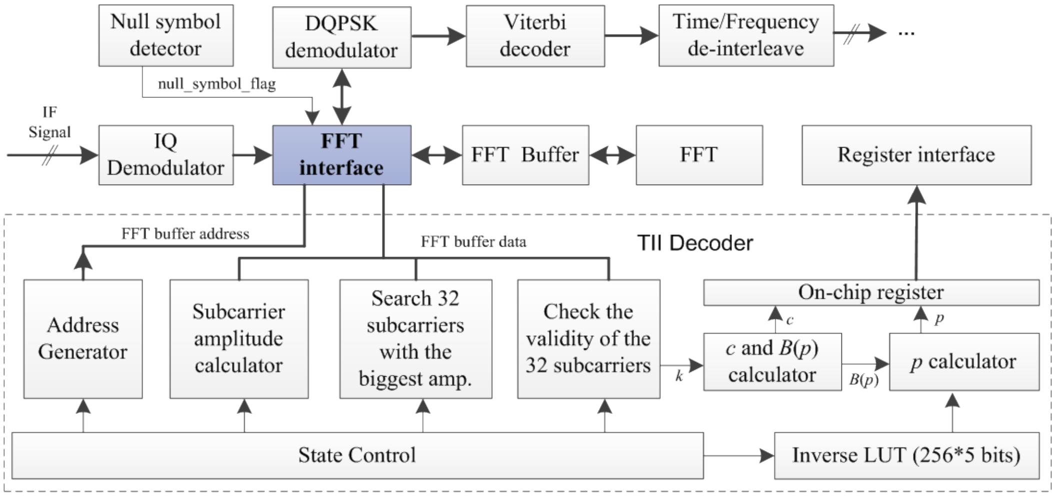

Fig. 4. The block diagram of the hardware TII decoder

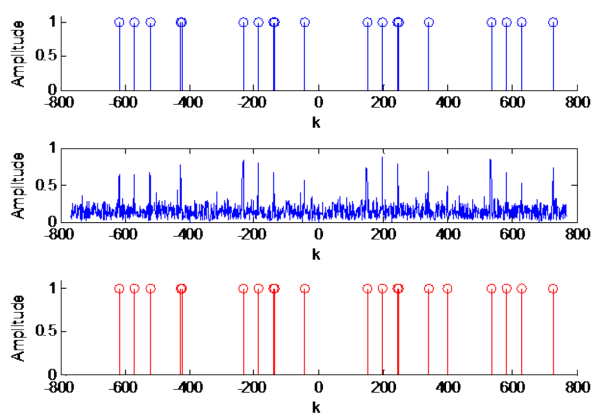

Fig. 5. The TII decoding result with SNR = 3dB

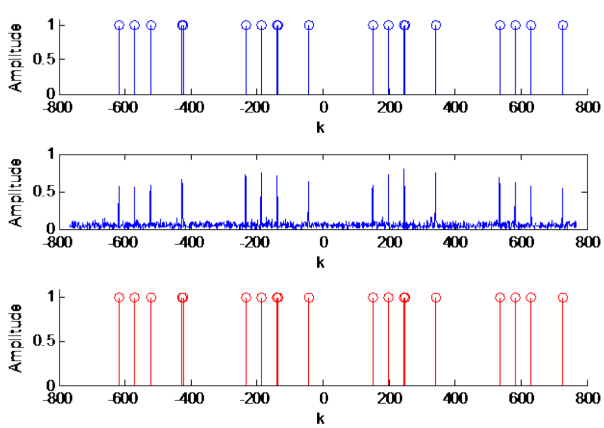

Fig. 6. The TII decoding result with SNR = 7dB

In order to check the efficiency of the TII decoder, we conducted 1000 tests for each SNR ranging from 3dB to 7.5dB in a step of 0.1dB. The test results are given in Fig. 7. It can be seen that for SNR higher than 6dB, the detection rate (the ratio of the number of the successful detections to the total detection times) is nearly 100%. Even for SNR = 3dB, which is a very bad receiving condition, the detection rate can reach to 50%. According to the DAB test standard [14], the required SNR is at least 6dB for an acceptable receiving quality, so the proposed TII decoder will be of good performance under all the working conditions of a DAB receiver. Although the TII decoder has failure detections, it will never give a wrong result because of the strict validity judgement conditions. This is proved by the experiment in which no wrong TII is reported even for SNR = 3dB, although there are nearly 50% failure detections.

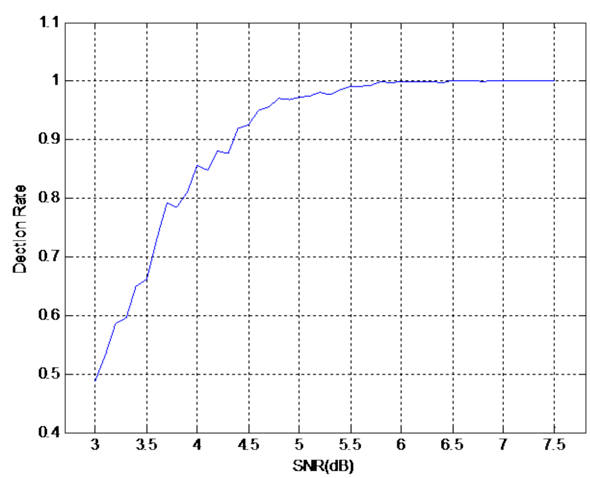

Fig. 7. The detection rates with SNR from 3 dB to 7.5 dB

The hardware TII decoder is designed using Verilog Hardware Description Language, and implemented in a Field Programmable Gate Array (FPGA) chip with only 286 logic elements and 1280 memory bits. The decoder is then integrated in a prototype DAB demodulator using the method stated in Section IV.B and Fig. 4 to receive the real DAB transmission signal. Test results show that it works correctly in the real-time conditions and the TII can be decoded within 1 second at most cases.

VI. Conclusion

In this paper, the TII coding theory of DAB and DMB is analyzed and a simple and reliable TII decoding algorithm is reported. The algorithm has only adding, compare, shift and addressing operations, making it suitable for both hardware and software implementations. Test results show that the algorithm achieves nearly 100% detection rate with a receiving SNR higher than 6dB. There are no wrong detections at all the test conditions, including the condition with SNR as low as 3dB. A software implementation method is proposed to enable the software-based DAB/DMB receivers to add the TII decoding functionality simply though firmware updating. A hardware TII decoder is also designed at a low cost of only 286 logic elements and 1280 memory bits, and with flexibility to be easily integrated in any DAB/DMB demodulator ASIC chips.

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