Adaptive Modulation Coding Overview and Learning

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Abstract: Adaptive modulation coding (AMC) is a technique that changes the modulation and coding modes according to the channel conditions. This paper introduces the principles and applications of adaptive modulation coding, mainly for beginners. By reading this paper, beginners can have a general concept of adaptive modulation coding relatively quickly. Meanwhile, this paper uses MATLAB to simulate the adaptive modulation coding based on QPSK and Turbo codes to improve the performance of BER decoding by limiting the number of iterations and BER requirements using adaptive modulation coding. And by comparing the two, the superiority of adaptive modulation coding compared with normal coding is ju dged, thus facilitating beginners to learn and understand.
Keywords: Adaptive Modulation Coding (AMC); MATLAB; QPSK; Turbo Codes; BER

1. Introduction

Adaptive modulation and Coding (AMC) is a technology that ADAPTS to change modulation and coding modes according to channel conditions.

As a simple example, in a system using AMC, users in the center of the cell are often assigned to higher order modulation or coding rates (64QAM, 3/4 bit rate Turbo code), while users at the edge of the cell are assigned to lower order modulation or coding rates (QPSK, 1/2 bit rate Turbo code).

The concept of "Adaptive modulation and coding" was put forward from the late 1960s to the early 1970s (J.F.Howes, 1968.FEB), but due to the limitations of hardware technology and lack of appropriate channel estimation algorithm , Adaptive modulation and coding did not attract much attention at that time.

However, with the development of technology and the progress of hardware equipment, the problems that originally hindered the adaptive modulation and coding system were gradually solved. In addition to the increasingly limited bandwidth resources, the adaptive modulation and coding technology has gradually attracted people's attention due to its own advantages.

2. Principle

The basic principle of adaptive modulation and coding (AMC) technology is that when the channel state changes, the transmitting terminal keeps the transmitting power constant, but the modulation and coding mode change adaptively with the channel state, so as to obtain the maximum throughput under different channel states. To put it simply, it is the mutual transformation between modulation methods, among which the modulation methods used are mainly BPSK\QPSK\8PSK\16APSK\32APSK\16QAM\64QAM\etc, like literature 2 and 3.In this paper, an adaptive modulation system of QPSK\16QAM\64QAM is proposed and its spectrum utilization is analyzed.

Also in literature 3,such modulation methods as BPSK\QPSK\8PSK\16APSK are adopted. In modern mobile communication, the system has various physical layer (PHY) transmission modes to choose from, and adaptive modulation coding is actually a reasonable choice of PHY transmission mode according to channel conditions.

However, with PHY mode 8 (64 QAM modulation, with a bit rate of 3/4 convolutional code encoding), although the system throughput is high at a high SNR, normal communication cannot be carried out when the channel is at a low SNR. The adaptive modulation coding technique achieves a good compromise between the reliability and stability of communication and the system throughput, and successfully solves the contradiction between the two. The choice of transmission mode is determined by the adaptive modulation coding algorithm adopted by the system.

3. Adaptive modulation and coding system model

According to the change of channel condition, the modulation of speed and power can be adaptively changed, and the lower average bit error rate and the higher spectral efficiency can be obtained. Among them, the selection of the appropriate modulation series and the algorithm to determine the power distribution are the core factors of adaptive modulation, which determine the performance of the system. At the same time, the code rate can be changed to further improve the transmission efficiency.

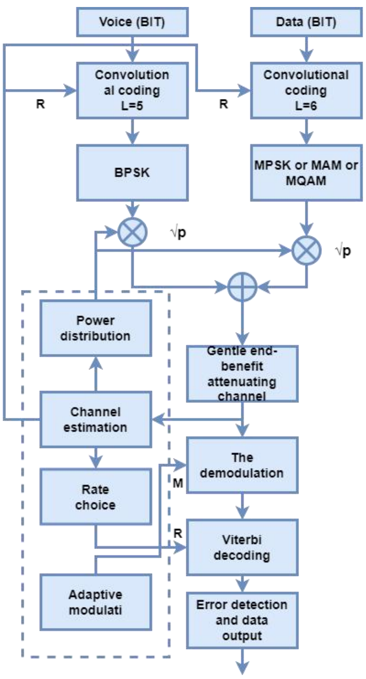


Figure 1 Flow chart of adaptive modulation and coding system

Figure 1 is a block diagram of adaptive modulation and coding for third-generation mobile communications. Inside the dotted line is the core part of the adaptive system. The receiving end makes channel estimation according to the received signals, determines the coding scheme, modulation scheme and corresponding modulation parameters, and then sends back the unified feedback channel (assuming that the delay of the feedback channel is negligible and the feedback is error-free) to the sending end, which makes corresponding adjustments for the next transmission. It is assumed that the channel is time-invariant in a short period of time. It is assumed that the speech is modulated in BPSK mode, convolutional encoding with constraint length L = 5 is adopted, and the data is modulated in MPASK, MAM or MQAM mode, and convolutional encoding with L = 6 is adopted, with orthogonal transmission of speech and data. Voice services need real-time transmission but have low requirements for BER, data services have low requirements for real-time but high requirements for bit error rate, which can be detected and re-sent. Therefore, the voice bit error rate threshold BERvth ≤ 10-3 and the data bit error rate threshold BERdth ≤ 10-4 can be set, and the modulation series M and bit rate R can be determined according to the relationship between bit error rate and noise ratio (CNR), and the power can be allocated.

4. Simulation of adaptive tuning to coding

The previous sections briefly introduce adaptive modulation coding and its principle. I will conduct comparative experiments in this section to explore the characteristics and performance improvement of adaptive modulation coding.

4.1 The simulation design

In the iteration process, the difference between the last iteration and the current iteration is judged to determine whether the iteration should be terminated. If the SNR is poor, the maximum number of iterations set by the iteration value will be output.

In the same code environment, the advantages of adaptive modulation coding are analyzed by comparing the differences between conventional modulation coding and adaptive modulation coding.

4.2 Turbo code

Turbo Code, also known as Parallel Concatenated Convolutional Code (PCCC, Parallel Concatenated Convolutional Code), ingeniously combines the Convolutional Code and the random interleaver, at the same time, the Convolutional Code is realized by the interleaver to construct the long Code from the short Code, and the soft-output iterative decoding is used to approximate the maximum likelihood decoding.

It can be seen that Turbo code charging utilizes the basic conditions of Shannon channel coding theorem, so the performance close to Shannon limit is obtained.

4.3 QPSK code

Quadrature Phase Shift Keying (QPSK) is a digital modulation method.It is divided into absolute phase shift and relative phase shift. Because of the phase fuzzy problem in absolute phase shift mode, relative phase shift mode DQPSK is mainly used in practice.

The sinusoidal carrier of PSK signal has four possible discrete phase states, and each carrier phase carries two binary symbols. The signal representation is as follows:

Si(t)= (+)

Between 0 and Ts, I is equal to 1,2,3,4. Where Ts is the quaternary symbol interval, (I =1,2,3,4) is the phase of the sinusoidal carrier, and there are four possible states.

4.4 Adaptive modulation coding based on QPSK and Turbo codes

```
clc
close all
clear all
g = [ 1 1 1; 1 0 1 ];
[n,K] = size(g);
m = K - 1;
nstates = 2^m;
puncture = 1; % 1 – the bit rate is 1/3 0 - 1/2
rate = 1 / (2 + puncture);
a = 1;
RUNS = 100; % Run number
L_total = 1024;
len_code = L_total * 3; % bit rate 1/3
ITER = 5; % Set the maximum number of Turbo decoding iterations, generally greater than or equal to 5
EbN0db = -4 : 1 : 8;
G_err_bit = zeros(1, length(EbN0db)); % Demodulation bit error rate Statistics
errs_bits = zeros(1, length(EbN0db)); % Decoding bit error rate statistics
iterNumStatic = zeros(1, length(EbN0db)); % Decoding bit error rate statistics
tic
for idB = 1 : length(EbN0db)
en = 10^(EbN0db(idB) / 10); % DB SNR conversion
L_c = 4 * a * en * rate; % Channel parameter coefficient
sigma = 1/sqrt(2 * en); % Standard Gaussian white noise variance
% Load the interleaver required for Turbo coding and decoding
load('interleaver1024.mat','alpha')
% The state transition relation of coding and decoding is established
[next_out, next_state, last_out, last_state] = trellis(g);
dec_bit = zeros(1,len_code);
for iRun = 1 : RUNS
% Generates a random data source for modulation coding src
= randn(1, L_total) > 0; % Turbo
en_output = encoderm(src, g, alpha, puncture) ; % encoder output
```

```
(+1/-1)
modSignal = (en_output(1 : 2 : end) + 1i * en_output(2 : 2 : end))/ sqrt(2);
% Add channel noise -white Gaussian noise channel noiseSignal = modSignal + sigma * (randn(1, length(modSignal)) + 1i * randn(1,
length(modSignal))) / sqrt(2);
% QPSK demodulation
decSignal(1 : 2 : len_code) = real(noiseSignal);
decSignal(2 : 2 : len_code) = imag(noiseSignal);
dec_bit = 2 * (decSignal(1 : len_code) > 0) - 1;
% Turbo decoding
[decBits, iterr] = TurboDecoder_adp(decSignal, ITER, g, alpha,
L_total, next_out, next_state, last_out, last_state);
% Demodulation error bit statistics
G_err_bit(idB) = G_err_bit(idB) + sum(dec_bit ~= en_output);
% Turbo decoding error bit statistics
errs_bits(idB) = errs_bits(idB) + sum(decBits ~= src);
% Iteration count
iterNumStatic(idB) = iterNumStatic(idB) + iterr;
end
% Modulation and demodulation bit error rate
G_err_bit(idB) = G_err_bit(idB) / (RUNS * len_code);
% Turbo decoding bit error rate
errs_bits(idB) = errs_bits(idB) / (RUNS * L_total);
end
toc
figure(1)
% plot(EbN0db, G_err_bit);
semilogy(EbN0db, G_err_bit)
title("QPSK demodulation bit error rate");
figure(2)
% plot(EbN0db, errs_bits);
semilogy(EbN0db, errs_bits)
title("Turbo decoding bit error rate ");
figure(3)
plot(EbN0db, iterNumStatic);
title("Turbo decoding iteration times ");
```

4.5 Conventional modulation coding based on QPSK and Turbo codes

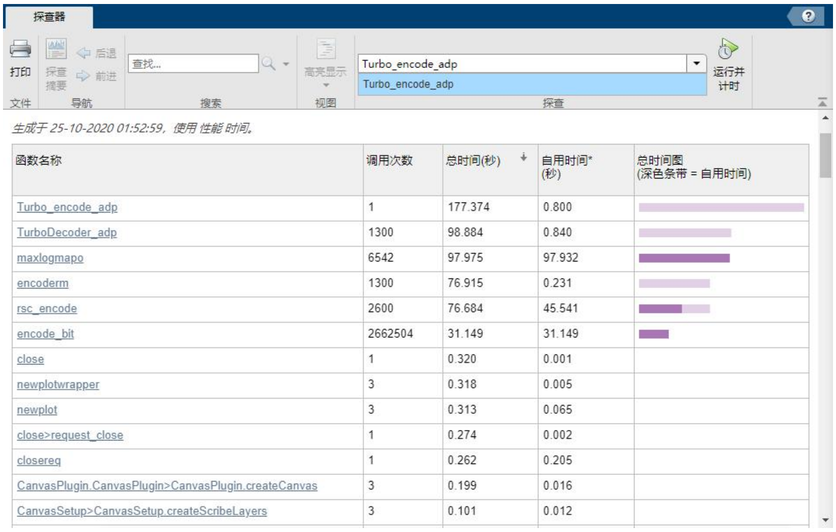
```
clc
close all
clear all
g = [ 1 1 1; 1 0 1 ];
[n,K] = size(g);
m = K - 1;
nstates = 2^m;
puncture = 1;
rate = 1 / (2 + puncture);
a = 1;
RUNS = 100;
L_total = 1024;
len_code = L_total * 3;
ITER = 5;
EbN0db = -4 : 1 : 8;
errs_bits = zeros(1, length(EbN0db));
iterNumStatic = zeros(1, length(EbN0db));
tic
for idB = 1 : length(EbN0db)
en = 10^(EbN0db(idB) / 10);
L_c = 4 * a * en * rate;
sigma = 1/sqrt(2 * en);
load('interleaver1024.mat','alpha')
[next_out, next_state, last_out, last_state] = trellis(g);
dec_bit = zeros(1,len_code);
for iRun = 1 : RUNS
src = randn(1, L_total) > 0;
en_output = encoderm(src, g, alpha, puncture) ; % encoder output (+1/-1)
modSignal = (en_output(1 : 2 : end) + 1i * en_output(2 : 2 : end)) / sqrt(2);
noiseSignal = modSignal + sigma * (randn(1, length(modSignal)) + 1irandn(1, length(modSignal))) / sqrt(2);
decSignal(1 : 2 : len_code) = real(noiseSignal);
decSignal(2 : 2 : len_code) = imag(noiseSignal);
dec_bit = 2 * (decSignal(1 : len_code) > 0) - 1;
```

```
decBits = TurboDecoder_com(decSignal, ITER, g, alpha, L_total,
next_out, next_state, last_out, last_state);
errs_bits(idB) = errs_bits(idB) + sum(decBits ~= src);
iterNumStatic(idB) = iterNumStatic(idB) + ITER;
end
errs_bits(idB) = errs_bits(idB) / (RUNS * L_total);
end
toc
figure(1)
% plot(EbN0db, errs_bits);
semilogy(EbN0db, errs_bits)
title("Turbo decoding bit error rate");
% Output iteration times is constant.
figure(2)
semilogy(EbN0db,iterNumStatic )
title("Turbo decoding iteration times");
```

4.6 Simulation results and comparison

4.6.1 Performance time

Adaptive modulation coding based on QPSK and Turbo codes



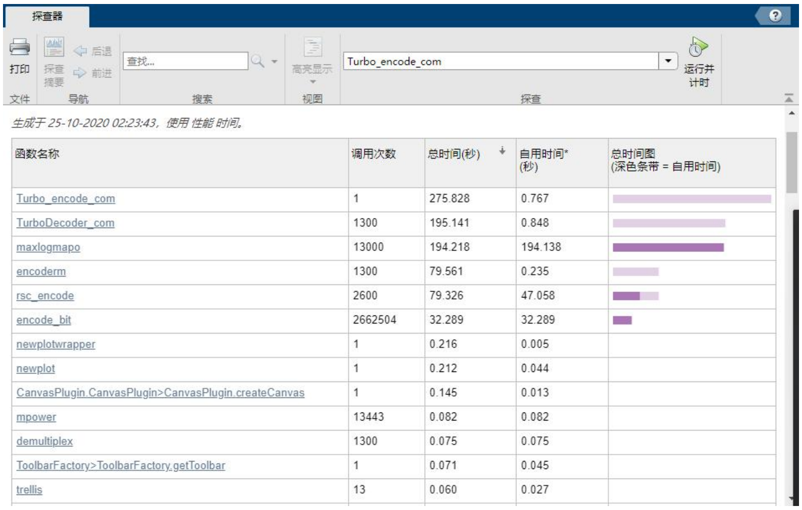
生成于 25-10-2020 01:52:59，使用 性能 时间。

函数名称	调用次数	总时间(秒)	自用时间*(秒)	总时间图(深色条带 = 自用时间)
Turbo_encode_adp	1	177.374	0.800	
TurboDecoder_adp	1300	98.884	0.840	
maxlogmapo	6542	97.975	97.932	
encoderm	1300	76.915	0.231	
rsc_encode	2600	76.684	45.541	
encode_bit	2662504	31.149	31.149	
close	1	0.320	0.001	
newplotwrapper	3	0.318	0.005	
newplot	3	0.313	0.065	
close>request_close	1	0.274	0.002	
closereq	1	0.262	0.205	
CanvasPlugin.CanvasPlugin>CanvasPlugin.createCanvas	3	0.199	0.016	
CanvasSetup>CanvasSetup.createScribeLayers	3	0.101	0.012	

Figure2 Adaptive modulation coding based on QPSK and Turbo codes

The total time is 177347s.

Conventional modulation coding based on QPSK and Turbo codes



函数名称	调用次数	总时间(秒)	自用时间*(秒)	总时间图(深色条带 = 自用时间)
Turbo_encode_com	1	275.828	0.767	
TurboDecoder_com	1300	195.141	0.848	
maxlogmapo	13000	194.218	194.138	
encoderm	1300	79.561	0.235	
rsc_encode	2600	79.326	47.058	
encode_bit	2662504	32.289	32.289	
newplotwrapper	1	0.216	0.005	
newplot	1	0.212	0.044	
CanvasPlugin.CanvasPlugin>CanvasPlugin.createCanvas	1	0.145	0.013	
mpower	13443	0.082	0.082	
demultiplex	1300	0.075	0.075	
ToolbarFactory>ToolbarFactory.getToolbar	1	0.071	0.045	
trellis	13	0.060	0.027	

Figure3 Conventional modulation coding based on QPSK and Turbo codes

The total time is 275828s.

4.6.2 QPSK demodulation bit error rate

Adaptive modulation coding based on QPSK and Turbo codes

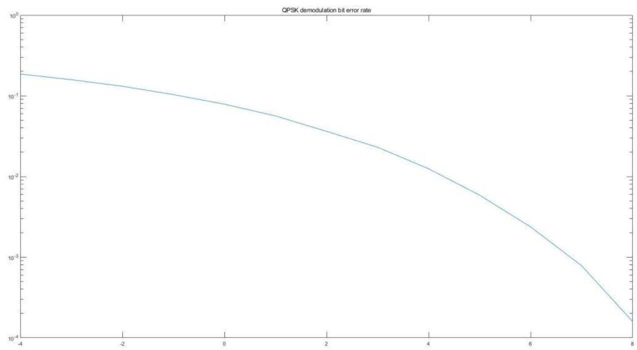


Figure4 QPSK demodulation decoding bit error rate

Conventional modulation coding based on QPSK and Turbo codes

It is same as Adaptive modulation coding based on QPSK and Turbo codes.

4.6.3 Turbo decoding bit error rate

Adaptive modulation coding based on QPSK and Turbo codes

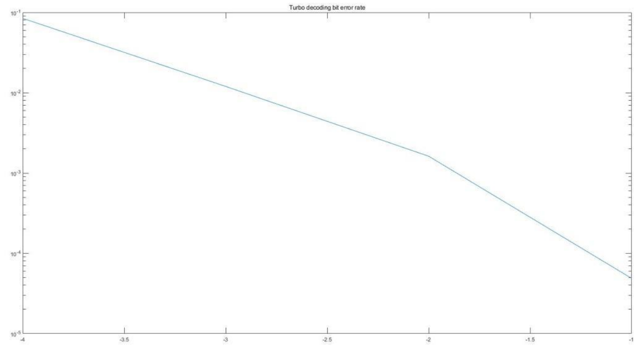


Figure5 Turbo decoding bit error rate

Conventional modulation coding based on QPSK and Turbo codes

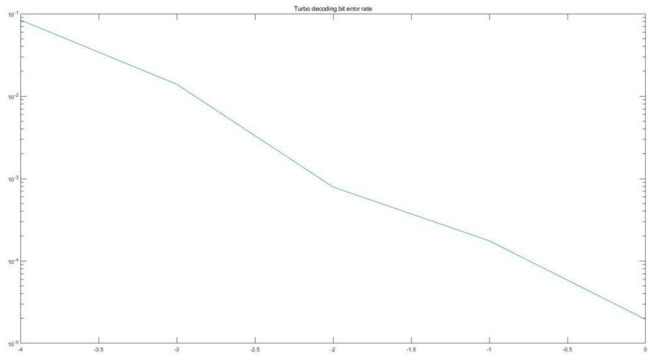


Figure6 Turbo decoding bit error rate

4.6.4 Turbo decoding iteration times

Adaptive modulation coding based on QPSK and Turbo codes

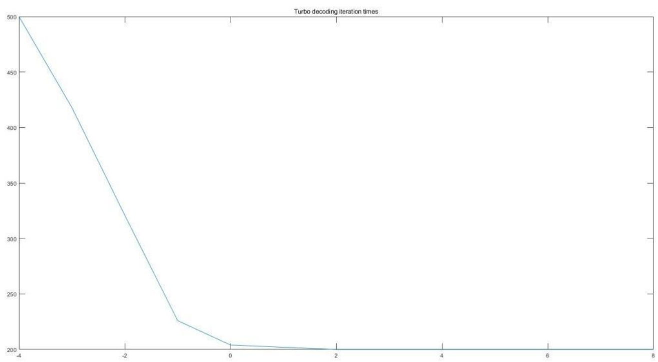


Figure7 Turbo decoding iteration times

Conventional modulation coding based on QPSK and Turbo codes

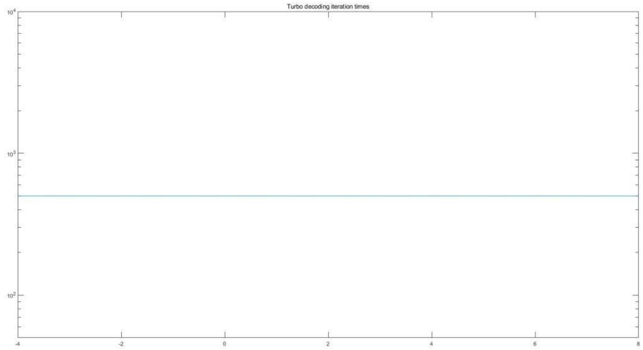


Figure8 Turbo decoding iteration times

4.7 Conclusion of simulation

By using the running time of MATLAB, assuming that the running time of MATLAB is the actual hardware running time, the running time of the program using adaptive response modulation coding is 177,347 seconds, which is far less than the 275,828 seconds of the conventional modulation coding. Therefore, we can conclude that adaptive modulation coding will bring greater performance improvement to the hardware. In the bit-error rate diagram comparing adaptive and conventional decoding, we can intuitively see that under the same conditions, the best information quality transmission is always adaptive modulation coding. Similarly, by comparing the variation diagram of iteration times with SNR and iteration times, the accuracy of adaptive modulation coding is improving continuously, while the conventional modulation system does not change and is always 500. Therefore, adaptive modulation is far superior to conventional modulation. Adaptive modulation coding is based on channel estimation and feedback. At one end of the transmitting information, the receiver's feedback channel state information or the channel state information obtained by the receiver's own channel estimation is precoded to eliminate the influence of channel and noise on the signal as much as possible, so as to greatly improve the correctness of the signal transmission.

5. The characteristics of adaptive modulation coding technology

Adaptive modulation coding technology in mobile communication system can not only counter the time variability of channel, but also overcome the influence of average path loss, slow fading and fast fading. Through research and simulation, I found that adaptive modulation coding technology has the following characteristics:

(1) The adaptive modulation coding techniques along with the change of channel environment change the rate of data transmission, can guarantee the data of fixed rate and time delay, so do not apply to the need of fixed data rate and delay circuit exchange of business, such as voice business, video phone business, applies only to the data rate and the delay did not require packet switching operations, such as the web browsing and file downloads.

(2) Adaptive coded modulation technology to keep the transmission power constant, letter condition good user has higher data transfer rate, poor channel conditions and the user can only communicate with low data rate, not only to avoid the "near-far effect" of the power control technique, but also overcome the

interference of the user to other user change problem, reduces the network interfered with allowance, solved the fast power control technology of "noise" effect should be, to increase the capacity of the system.

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