高等学校教材·电子信息

信息与通信工程专业科技英语

王朔中 主编

清华大学出版社

高等学校教材·电子信息

信息与通信工程
专业科技英语

王朔中　主编

清华大学出版社
北　京

内 容 简 介

本书摒弃陈旧的教学模式，强调将英语作为获取专业知识和有关信息、进行交流的工具，着重培养学生以较高准确性和足够的速度阅读专业资料和文献的能力，兼顾一定的专业英语表达能力。

本书共 20 单元，每一单元包括课文、词汇、难点注释、课外阅读资料、语言知识、习题。课文和阅读资料的取材涉及信息与通信工程及相关领域的基础知识和新技术进展，包括了信息科学、通信工程、信号处理、电子技术、生物医学工程、计算机应用等有关领域的英语科技文章和技术资料，并注意较广的科技英语基本词汇和适当的语言难度。教学中可根据课文内容有选择地用英语讲述某些专业基础知识，在学习专业英语同时扩大知识面。

本书可作为高等院校信息、通信、电子类专业本科和研究生的科技英语教材。

图书在版编目（CIP）数据

信息与通信工程专业科技英语/王朔中主编. —北京：清华大学出版社，2004.9
（高等学校教材·电子信息）
ISBN 7-302-09293-1

Ⅰ. 信…　Ⅱ. 王…　Ⅲ. ①信息技术–英语–高等学校–教材 ②通信工程–英语–高等学校–教材　Ⅳ. H31

中国版本图书馆 CIP 数据核字（2004）第 086081 号

出 版 者：清华大学出版社　　**地　　址**：北京清华大学学研大厦
http://www.tup.com.cn　　**邮　　编**：100084
社总机：010-62770175　　**客户服务**：010-62776969
责任编辑：魏江江
封面设计：杨　兮
印 装 者：北京鑫海金澳胶印有限公司
发 行 者：新华书店总店北京发行所
开　　本：186×260　**印张**：18.5　**字数**：447 千字
版　　次：2004 年 9 第 1 版　2004 年 9 月第 1 次印刷
书　　号：ISBN 7-302-09293-1/TP · 6516
印　　数：1~5000
定　　价：26.00 元

本书如存在文字不清、漏印以及缺页、倒页、脱页等印装质量问题，请与清华大学出版社出版部联系调换。联系电话：（010）62770175-3103 或（010）62795704

高等学校教材·电子信息
编审委员会

曾凡鑫　（重庆通信学院　教授）
曾喆昭　（长沙理工大学电气与信息工程学院　教授）
曾孝平　（重庆大学通信工程学院　教授）
彭启琮　（电子科技大学　教授）
谢显中　（重庆邮电学院　教授）
樊昌信　（西安电子科技大学通信工程学院　教授）

出版说明

改革开放以来，特别是党的十五大以来，我国教育事业取得了举世瞩目的辉煌成就，高等教育实现了历史性的跨越，已由精英教育阶段进入国际公认的大众化教育阶段。在质量不断提高的基础上，高等教育规模取得如此快速的发展，创造了世界教育发展史上的奇迹。当前，教育工作既面临着千载难逢的良好机遇，同时也面临着前所未有的严峻挑战。社会不断增长的高等教育需求同教育供给特别是优质教育供给不足的矛盾，是现阶段教育发展面临的基本矛盾。

教育部一直十分重视高等教育质量工作。2001 年 8 月，教育部下发了《关于加强高等学校本科教学工作，提高教学质量的若干意见》，提出了十二条加强本科教学工作提高教学质量的措施和意见。2003 年 6 月和 2004 年 2 月，教育部分别下发了《关于启动高等学校教学质量与教学改革工程精品课程建设工作的通知》和《教育部实施精品课程建设提高高校教学质量和人才培养质量》文件，指出"高等学校教学质量和教学改革工程"，是教育部正在制订的《2003—2007 年教育振兴行动计划》的重要组成部分，精品课程建设是"质量工程"的重要内容之一，教育部计划用五年时间(2003—2007 年)建设 1500 门国家级精品课程，利用现代化的教育信息技术手段将精品课程的相关内容上网并免费开放，以实现优质教学资源共享，提高高等学校教学质量和人才培养质量。

为了深入贯彻落实教育部《关于加强高等学校本科教学工作，提高教学质量的若干意见》精神，紧密配合教育部已经启动的"高等学校教学质量与教学改革工程精品课程建设工作"，在有关专家、教授的倡议和有关部门的大力支持下，我们组织并成立了"清华大学出版社教材编审委员会"(以下简称"编委会")，旨在配合教育部制定精品课程教材的出版规划，讨论并实施精品课程教材的编写与出版工作。"编委会"成员皆来自全国各类高等学校教学与科研第一线的骨干教师，其中许多教师为各校相关院、系主管教学的院长或系主任。

按照教育部的要求，"编委会"一致认为，精品课程的建设工作从开始就要坚持高标准、严要求，处于一个比较高的起点上；精品课程教材应该能够反映各高校教学改革与课程建设的需要，要有特色风格、有创新性(新体系、新内容、新手段、新思路，教材的内容体系有较高的科学创新、技术创新和理念创新的含量)、先进性(对原有的学科体系有实质性的改革和发展，顺应并符合新世纪教学发展的规律，代表并引领课程发展的趋势和方向)、示范性(教材所体现的课程体系具有较广泛的辐射性和示范性)和一定的前瞻性。教材由个人申报或各校推荐(通过所在高校的"编委会"成员推荐)，经"编委会"认真评审，最后由清华大学出版社审定出版。

目前，针对计算机类和电子信息类相关专业成立了两个"编委会"，即"清华大学出版社计算机教材编审委员会"和"清华大学出版社电子信息教材编审委员会"。首批推出的特色精品教材包括以下三个系列：

(1) 高等学校教材·计算机应用——高等学校各类专业，特别是非计算机专业的计算机应用类教材。

（2）高等学校教材·计算机科学与技术——高等学校计算机相关专业的教材。

（3）高等学校教材·电子信息——高等学校电子信息相关专业的教材。

清华大学出版社经过近二十年的努力，在教材尤其是计算机和电子信息类专业教材出版方面树立了权威品牌，为我国的高等教育事业做出了重要贡献。清华版教材经过二十多年的精雕细刻，形成了技术准确、内容严谨的独特风格，这种风格将延续并反映在特色精品教材的建设中。

总策划　李家强

策　划　卢先和　丁　岭

清华大学出版社教材编审委员会

E-mail：dingl@tup.tsinghua.edu.cn

luxh@tup.tsinghua.edu.cn

前　　言

本教材是为信息技术、通信工程、电子工程以及其他有关专业的本科生、研究生学习科技英语而编写的。

英语作为一种交流工具对学生今后的发展十分重要。学生们毕业以后将要接触大量的英语资料，包括科学论文、技术文件、产品说明、商业信息等，有的还会在工作和学习中以英语为主要语言媒体。不少人从小学开始学了十几年英语，但是到了大学高年级仍然缺乏有效运用英语获取信息和知识的能力，更不要说以口头或书面方式来表达自己的思想、介绍科技成果了。这里的问题并不在于语法和词汇。实际上许多学生已经具备了丰富的英语知识，他们熟知语法规则和大量单词、善于应对各类考试，但这不等于他们已经掌握了英语。关于英语的知识并不是英语本身。到了需要实际运用英语进行交流的时候，不少人发现他们的阅读能力远远不能满足要求，也不知道如何正确地用英语写作。

鉴于以上情况，本书强调英语的实际运用而不是语法。考虑到课堂教学时间的有限性和大部分学生的实际需要，我们将重点放在科技文章的阅读上。全书供 20 个单元，每一单元包括课文、词汇、注释、语法知识或翻译写作要点、课外阅读材料、练习六个部分。课文和课外阅读内容涉及通信技术、信号与信息处理、电子线路与系统、微波、光纤、生物医学工程、计算机等领域。练习与课文内容没有直接联系，其目的不是为了复习课文或语法，而是作为阅读训练的补充。

必须指出，仅有课堂教学而缺少大量的阅读实践是无法真正突破英语阅读关的。有些人虽然能够借助词典进行阅读，但是理解的准确性差，而且阅读速度很慢，不能在一定的时间里获取足够多的信息，吞吐量太小。对他们来说，学了多年的英语还没有成为有效的交流工具，最多不过是偶尔借用一下的拐杖罢了。造成这种情况的原因之一就是阅读太少，因此我们主张广泛课外阅读，不能受教科书的限制。另外还应当结合专业学习和科研实践进行大量的阅读，有条件也可以看一些非科技类的英文读物。

英语写作不是本书的主要内容，书中仅在第 15~20 单元包括了作者关于科技英语写作的一些体会。我们认为除了专门的写作训练和必要的写作实践以外，大量阅读以及在阅读中留心观察、认真思考的良好习惯对于写作能力的提高也是至关重要的。熟读唐诗三百首，不会作诗也会吟，就是这个道理，当然不是机械地读，要动脑筋。

本书由王朔中主编，石海、黄素娟参加编写。参加编写工作的还有朱秋煜、石旭利、李颖洁，他们搜集整理了大量资料，提出了有益的建议，并在教材试用期间参与讨论，对部分书稿进行了校对并提出了修改意见。

因作者水平所限，书中差错和不当之处在所难免，敬请读者不吝指正。

作者 E-mail：shuowang@staff.shu.edu.cn

王朔中

目　　录

Unit 1

Digital Audio Compression Standard: AC3*

Text

Foreword

The United States Advanced Television Systems Committee (ATSC) was formed by the member organizations of the Joint Committee on InterSociety Coordination (JCIC)**, recognizing that the prompt, efficient and effective development of a coordinated set of national standards is essential to the future development of domestic television services.

One of the activities of the ATSC is exploring the need for and, where appropriate, coordinating the development of voluntary national technical standards for Advanced Television Systems (ATV).[1] The ATSC Executive Committee assigned the work of documenting the U.S. ATV standard to a number of specialist groups working under the Technology Group on Distribution (T3). The Audio Specialist Group (T3/S7) was charged with documenting the ATV audio standard.

This document was prepared initially by the Audio Specialist Group as part of its efforts to document the United States Advanced Television broadcast standard. It was approved by the Technology Group on Distribution on September 26, 1994, and by the full ATSC Membership as an ATSC Standard on November 10, 1994. Annex A, "AC-3 Elementary Streams in an MPEG-2 Multiplex," was approved by the Technology Group on Distribution on February 23, 1995, and by the full ATSC Membership on April 12, 1995. Annex B, "AC-3 Data Stream in IEC958 Interface," and Annex C, "AC-3 Karaoke Mode," were approved by the Technology Group on

---

* The United States Advanced Television Systems Committee, December 20, 1995

** The JCIC is presently composed of: the Electronic Industries Association (EIA), the Institute of Electrical and Electronic Engineers (IEEE), the National Association of Broadcasters (NAB), the Natioal Cable Television Association (NCTA), and the Society of Motion Picture and Television Engineers (SMPTE).

NOTE: The user's attention is called to the possibility that compliance with this standard may require use of an invention covered by patent rights. By publication of this standard, no position is taken with respect to the validity of this claim, or of any patent rights in connection therewith. The patent holder has, however, filed a statement of willingness to grant a license under these rights onreasonable and nondiscriminatory terms and conditions to applicants desiring to obtain such a license. Details may be obtained from the publisher.

Distribution on October 24, 1995, and by the full ATSC Membership on December 20, 1995. ATSC Standard A/53, *Digital Television Standard for HDTV Transmission*, references this document and describes how the audio coding algorithm described herein is applied in the U.S. ATV standard.

At the time of release of this document, the system description contained herein had not been verified by the transmission of signals from independently developed encoders to separately developed decoders.

Introduction

Motivation

In order to more efficiently broadcast or record audio signals, the amount of information required to represent the audio signals may be reduced. In the case of digital audio signals, the amount of digital information needed to accurately reproduce the original pulse code modulation (PCM) samples may be reduced by applying a digital compression algorithm, resulting in a digitally compressed representation of the original signal.[2] (The term compression used in this context means the compression of the amount of digital information which must be stored or recorded, and not the compression of dynamic range of the audio signal.) The goal of the digital compression algorithm is to produce a digital representation of an audio signal which, when decoded and reproduced, sounds the same as the original signal, while using a minimum of digital information (bitrate) for the compressed (or encoded) representation. The AC-3 digital compression algorithm specified in this document can encode from 1 to 5.1 channels of source audio from a PCM representation into a serial bit stream at data rates ranging from 32 kbps to 640 kbps. The 0.1 channel refers to a fractional bandwidth channel intended to convey only low frequency (subwoofer) signals.

A typical application of the algorithm is shown in Figure 1.1. In this example, a 5.1 channel audio program is converted from a PCM representation requiring more than 5 Mbps (6 channels $\times$ 48 kHz $\times$ 18 bits = 5.184 Mbps) into a 384 kbps serial bit stream by the AC-3 encoder. Satellite transmission equipment converts this bit stream to an RF transmission which is directed to a satellite transponder. The amount of bandwidth and power required by the transmission has been reduced by more than a factor of 13 by the AC-3 digital compression. The signal received from the satellite is demodulated back into the 384 kbps serial bit stream, and decoded by the AC-3 decoder. The result is the original 5.1 channel audio program.

Digital compression of audio is useful wherever there is an economic benefit to be obtained by reducing the amount of digital information required to represent the audio. Typical applications are in satellite or terrestrial audio broadcasting, delivery of audio over metallic or optical cables, or storage of audio on magnetic, optical, semiconductor, or other storage media.

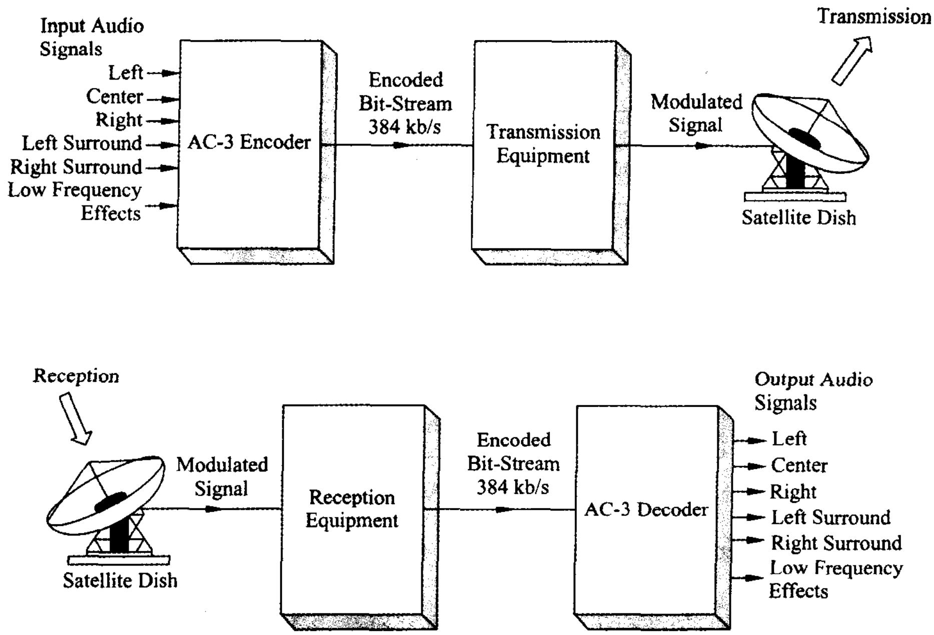

Figure 1.1 Example application of AC-3 to satellite audio transmission

Encoding

The AC-3 encoder accepts PCM audio and produces an encoded bit stream consistent with this standard. The specifics of the audio encoding process are not normative requirements of this standard. Nevertheless, the encoder must produce a bit stream matching the syntax described in Section 5, which, when decoded according to Sections 6 and 7, produces audio of sufficient quality for the intended application. Section 8 contains informative information on the encoding process. The encoding process is briefly described below.

The AC-3 algorithm achieves high coding gain (the ratio of the input bit-rate to the output bit-rate) by coarsely quantizing a frequency domain representation of the audio signal. A block diagram of this process is shown in Figure 1.2. The first step in the encoding process is to transform the representation of audio from a sequence of PCM time samples into a sequence of blocks of frequency coefficients. This is done in the analysis filter bank. Overlapping blocks of 512 time samples are multiplied by a time window and transformed into the frequency domain. Due to the overlapping blocks, each PCM input sample is represented in two sequential transformed blocks. The frequency domain representation may then be decimated by a factor of two so that each block contains 256 frequency coefficients. The individual frequency coefficients are represented in binary exponential notation as a binary exponent and a mantissa.[3] The set of exponents is encoded into a coarse representation of the signal spectrum which is referred to as the spectral envelope. This spectral envelope is used by the core bit allocation routine which determines how many bits to use to encode each individual mantissa. The spectral envelope and the coarsely quantized mantissas for 6 audio blocks (1536 audio samples) are formatted into an AC-3 frame. The AC-3 bit stream is a sequence of AC-3 frames.

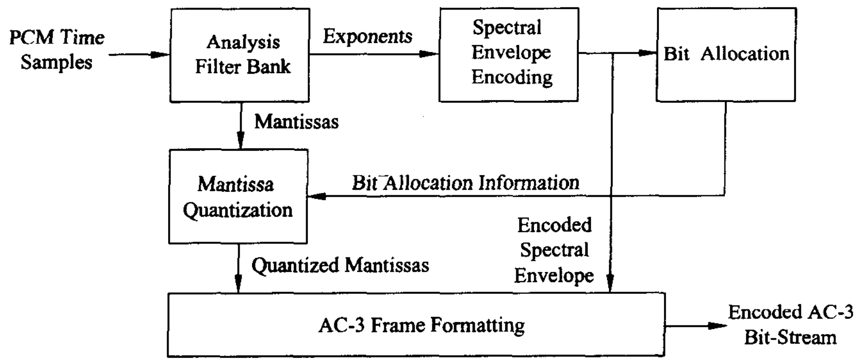

Figure 1.2 The AC-3 encoder

The actual AC-3 encoder is more complex than indicated in Figure 1.2. The following functions not shown above are also included.

1. A frame header is attached which contains information (bit-rate, sample rate, number of encoded channels, etc.) required to synchronize to and decode the encoded bit stream.[4]
2. Error detection codes are inserted in order to allow the decoder to verify that a received frame of data is error free.
3. The analysis filter bank spectral resolution may be dynamically altered so as to better match the time/frequency characteristic of each audio block.[5]
4. The spectral envelope may be encoded with variable time/frequency resolution.
5. A more complex bit allocation may be performed, and parameters of the core bit allocation routine modified so as to produce a more optimum bit allocation.
6. The channels may be coupled together at high frequencies in order to achieve higher coding gain for operation at lower bit-rates.
7. In the two-channel mode a rematrixing process may be selectively performed in order to provide additional coding gain, and to allow improved results to be obtained in the event that the two-channel signal is decoded with a matrix surround decoder.[6]

Decoding

The decoding process is basically the inverse of the encoding process. The decoder, shown in Figure 1.3, must synchronize to the encoded bit stream, check for errors, and de-format the various types of data such as the encoded spectral envelope and the quantized mantissas. The bit allocation routine is run and the results used to unpack and de-quantize the mantissas. The spectral envelope is decoded to produce the exponents. The exponents and mantissas are transformed back into the time domain to produce the decoded PCM time samples.

The actual AC-3 decoder is more complex than indicated in Figure 1.3. The following functions not shown above are included.