Technical English

For Information and Communication Engineering

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开场白：好的英语是“读”出来的

学习任何语言都要大量阅读。“熟读唐诗三百首，不会作诗也会吟”。大凡读得多的人，语言表达能力都比较强。

学英语也要大量阅读。英语好的人，一般说来都读过相当数量的书。只有读得多的人才能真正学好英语。

很多人热衷于做题目，以为这是学英语的正确途径。做的试题一套又一套，英语水平却并无实质性提高。

好的英语是“读”出来的，不是做练习“做”出来的。“读”出来的英语是地道的，自然的，因为学的是“语感”。“做”出来的英语往往流于生硬，因为是依赖“语法规则”硬做的。

节引自：黄源深，英语阅读也轻松，文汇报，2004.8.2

好的英语是“读”出来的（续）

我主张一口气读下去，即便有个别单词挡道，只要不影响整体内容的理解，就不必停下来查词典，因为那样会打断思路，影响阅读速度，甚至扼杀阅读兴趣。

阅读主要目的在于语言吸收上的潜移默化，在于获得语感。阅读需要“量”，没有大量语言“输入”难以学好英语。国内学习者恰恰语言输入量太少，输出量就更少，往往事倍功半。

练习要做，但要适量；重要的是阅读，大量的阅读。

节引自：黄源深，英语阅读也轻松，文汇报，2004.8.2

Unit One

Digital Audio Compression

概述

- 课文性质：技术文件。
- 特点：机构名称较多，造成句子较长。
  - 了解机构名称的缩写
  - 专业名词缩写
- 学会抓住长句要点。
- 了解技术文件中的某些正式表达方式。
- 结合课文初步了解音频编解码原理。

coordination	协调
voluntary	自发的，自愿的
annex	附件
multiplex	多样的，多路复用
herein	在此，如此
motivation	动机，刺激，推动力
algorithm	算法
representation	表达方式，表达
dynamic range	动态范围
bitrate	比特率
fractional	部分的，分数的
woofer	低音喇叭
transponder	应答器，转发器
demodulate	解调
terrestrial	地面的，地球上的
consistent	一致的
normative	规范的，标准的

syntax	句法
informative	提供信息的
decimate	抽取
coefficient	系数
exponent	指数
mantissa	尾数
envelope	包络
allocation	分配，指定
synchronize	使同步，同时发生
resolution	分辨率
parameter	参数
inverse	反转的，逆
unpack	解开
conceal	隐藏
mute	无声，使无声
rematrix	重新进行矩阵变换
dematrix	求矩阵反变换

迅速有效地制定一套相互协调的国家标准

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.

对于美国电视业今后的发展

在适当的情况下

寻求 … 的需要

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]

非强制性国家技术标准

协调 … 的开发工作

起草美国ATV标准的工作

The <u>ATSC Executive Committee</u> assigned the work of documenting the U.S. ATV standard to a number of specialist groups working under the <u>Technology Group on Distribution</u> (T3). The <u>Audio Specialist Group</u> (T3/S7) was charged with documenting the ATV audio standard.

"发布技术委员会"（T3）下设的一系列专家组

作为制定美国ATV广播标准的一部分

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.

ATSC全体成员

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 Distribution on October 24, 1995 and by the full ATSC Membership on December 20, 1995.

引用了本文件

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.

叙述了本文件所述的音频编码算法是如何用于美国ATV标准的

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.

由独立开发的编码器到分别开发的解码器之间进行传输

Motivation

用于表示音频的信息量

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]

用于精确重建原始脉冲编码调制样本所需要的数字信息量

由此产生原信号的数字压缩形式

在这里

(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.)

压缩必须存储或记录的数字信息量

给出与原始信号相同的听觉效果

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.

其压缩（或编码）形式使用最少的数字信息量（比特率）

PCM形式音频源信号

从1至5.1通道的音频源

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.

从 … 编码为32kbps至640bps的串行比特流

需要5Mbps以上码率的PCM形式
（6通道×48kHz×18比特＝5.184Mbps）

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×48 kHz×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.

送到卫星收发器

从 … 转换成384kbps码率的比特流

下降了13倍以上

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.

解调回到384kbps的串行比特流

只要 … 具有经济效益

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.

经过电缆或光缆传输音频，或在磁、光、半导体或其他存储介质上存储音频信号

符合本标准的

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.

对于具体应用产生音质足够好的音频信号

通过对音频信号的频域形式进行粗量化

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.

将音频信号从一系列PCM时域样本形式转换为一系列频率系数的块

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.

被乘以一个时间窗函数

以因子2抽取（频域样本）

表示在两个相连的频域块中

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]

用二进制指数形式表示，有一个<u>二进制指数和一个尾数</u>

信号频谱的粗放表示形式

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.

对每一尾数编码时所需要的比特数

与编码比特流实现同步并将它解码

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]

Error detection codes are inserted in order to allow the decoder to verify that a received frame of data is error free.

没有误码

更好地匹配每一音频信号块的时频特性

The analysis filter bank spectral resolution may be dynamically altered so as to better match the time/frequency characteristic of each audio block.[5]

The spectral envelope may be encoded with variable time/frequency resolution.

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.

可以进行更复杂的比特分配，可以修改核心比特分配例程的参数，以生成更优的比特匹配。

在低比特率工作时实现更高的编码增益

The channels may be coupled together at high frequencies in order to achieve higher coding gain for operation at lower bit-rates.

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]

在使用矩阵环绕声解码器解码双声道信号时

重新进行矩阵变换

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.

与编码的比特流同步，检查有无误码，将各种数据如编码的频谱包络和量化的尾数解格式

比特分配例程

省略“are”

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.

当各通道被重新进行矩阵变换时

检测到误码时

Error concealment or muting may be applied in case a data error is detected.

Channels which have had their high-frequency content coupled together must be decoupled.

Dematrixing must be applied (in the 2-channel mode) whenever the channels have been rematrixed.

The synthesis filter bank resolution must be dynamically altered in the same manner as the encoder analysis filter bank had been during the encoding process.

以编码时分析滤波器所用的同样方式

Footnote of p.1

- 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.
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Footnote of p.1 (cont.)

- **The patent holder has, however, filed a statement of willingness to grant a license under these rights on reasonable and nondiscriminatory terms and conditions to applicants desiring to obtain such a license. Details may be obtained from the publisher.**
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结合课文的思考题

- **What is the difference between the compression of signal dynamic range and the compression discussed in this text?**
- **Describe the basic procedures involved in the AC3 encoding.**

Exercises

- **Without data reduction, digital audio signals typically consist of 16 bit samples recorded at a sampling rate more than twice the actual audio bandwidth (e.g., 44.1 kHz for Compact Disks). So you end up with more than *1.4 Mbit* to represent just *one second of stereo music in CD quality*.**
- 如不进行数据压缩，数字音频信号由高于实际音频带宽两倍的采样频率（例如**CD**的**44.1kHz**）记录的**16**比特样本组成。因而你所得到的是以**1.4**兆比特以上的数据表示**1**秒钟**CD**质量的立体声音乐。

Exercises

- **By using MPEG audio coding, you may shrink down the original sound data from a CD by a factor of 12, without losing sound quality. Factors of 24 and even more still maintain a sound quality that is significantly better than what you get by just reducing the sampling rate and the resolution of your samples. Basically, this is realized by *perceptual coding* techniques addressing the perception of sound waves by the human ear.**
- 应用**MPEG**音频编码技术，你可以将**CD**的原声数据压缩**12**倍而不损失音质。压缩**24**倍以至更多仍可保持音质大大优于简单降低采样频率和样本分辨率所得到的结果。原则上这是通过感知编码技术实现的，这种技术是关于人耳对声波感知特性的。

Exercises

- **By exploiting stereo effects and by limiting the audio bandwidth, the coding schemes may achieve an acceptable sound quality at even lower bitrates. MPEG Layer-3 is the most powerful member of the MPEG audio coding family. For a given sound quality level, it requires the lowest bitrate; or for a given bitrate, it achieves the highest sound quality.**
- 编码方案通过利用立体声效应并限制音频带宽能以更低的比特率实现可接受的音质。**MP3**是**MPEG**音频编码系列中最强有力的一种。对于给定的音质水平它要求的比特率最低；或者对于给定的比特率它能实现最佳音质。

Exercises

- **In all international listening tests, MPEG Layer-3 impressively proved its superior performance, maintaining the original sound quality at a data reduction of 1:12 (around 64 kbit/s per audio channel). If applications may tolerate a limited bandwidth of around 10 kHz, a reasonable sound quality for stereo signals can be achieved even at a reduction of 1:24.**
- 在所有国际听力测试中，**MP3**引人注目地证明了它的优越性能：以**1:12**的压缩率（每一声道约**64kbps**）保持原始音质。假如某些实际应用仅容忍约**10kHz**的有限带宽，即使在**1:24**的压缩率还是可以得到尚满意的立体声音质。

Exercises

- **Early in the twentieth century, it was found that light could cause atoms to emit electrons and that, when light released an electron from an atom, the energy possessed by the electron very greatly exceeded that which the atom could, according to the electromagnetic wave theory, have received.**

  在二十世纪早期，人们发现光能够使原子放出电子，而且当光从原子中释放一个电子时，电子所包含的能量大大超过由电磁波理论得到的原子所接收的能量。

Exercises

- **However, in a properly designed DC amplifier the effect of transistor parameter variation, other than Ico, may be practically eliminated if the operation point of each stage is adjusted so that it remains in the linear operation range of the transistor as temperature varies.**

  然而在设计得当的直流放大器中，若调节每一级的工作点使之在温度变化时保持在晶体管线性区，就能在实际上消除**Ico**以外的晶体管参数变化所造成的影响。

Exercises

- **In a television scanning generator using a pair of free-running relaxation oscillators, free-running frequencies of the oscillators are set slightly below the horizontal and vertical pulse rates, and the stripped pulses are used to trigger the oscillators prematurely and thus to synchronize them to the line and half-frame rates.**

在采用一对自由张弛振荡器的电视扫描发生器中，振荡器的自由频率被设置得略低于水平和垂直（扫描）频率，分离出来的脉冲被用来提前触发振荡器从而使它们与行频和半帧频（场频）同步。