



大模型技术30讲（英文&中文批注）

大模型技术30讲（原版），30 Essential Questions and Answers on Machine Learning and AI

1.背景

买了一本《大模型技术30讲》，简单阅读了下，要点突出，对于入门、加深关键点理解，很有用。

但是，也存在问题 & 未满足的诉求：

- 1.《大模型技术30讲》纸质版（2025年3月第2次印刷），印刷质量，偏差；
- 2.大部分 **术语**，都翻译为中文，不利于中英对比，特别是 AI 领域基本都是英文的，需要我们熟悉 **英文术语**。
- 3.希望有 **电子版** 资料，方便利用 AI 工具辅助理解，提升效率。

因此，找到 [原始文档：Machine Learning Q and AI](#)，反复消化原始信息。

然后，将原始信息，转录为 md 格式，并且，存储在 github 上。

2.项目介绍

本项目是系统性学习大模型技术要点的教程，基于原始的《30 Essential Questions and Answers on Machine Learning and AI》，进行中文批注，增强可读性。

2.1.你将收获什么？

1. 系统性学习：大模型技术要点的教程
2. 熟悉核心的术语（英文 + 中文）
3. 共同维护，中文批注，增强可读性，贡献给开源社区



2.2.迭代计划

1. 首版：中文标注:
2. 导出 pdf 文件:
3. 同步到多个开源社区: TODO

3.在线阅读

在线阅读: [《大模型技术30讲》](#) , pdf 文件: [《大模型技术30讲-PDF版本》](#)

深入理解 LLM 核心原理, 直击要点



4.如何贡献

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- 💡 功能建议 - 有好想法就告诉我们
- 📝 内容完善 - 帮助改进教程内容

Note: 后续资料会附上贡献者名单.

30 Essential Questions and Answers on Machine Learning and AI

Machine learning and AI are moving at a rapid pace. Researchers and practitioners are constantly struggling to keep up with the breadth of concepts and techniques. This book provides bite-sized bits of knowledge for your journey from machine learning beginner to expert, covering topics from various machine learning areas. Even experienced machine learning researchers and practitioners will encounter something new that they can add to their arsenal of techniques.



What People Are Saying

"Sebastian has a gift for distilling complex, AI-related topics into practical takeaways that can be understood by anyone. His new book, Machine Learning Q and AI, is another great resource for AI practitioners of any level." "Cameron R. Wolfe, Writer of Deep (Learning) Focus

"Sebastian uniquely combines academic depth, engineering agility, and the ability to demystify complex ideas. He can go deep into any theoretical topics, experiment to validate new ideas, then explain them all to you in simple words. If you're starting your journey into machine learning, Sebastian is your guide." "Chip Huyen, Author of Designing Machine Learning Systems

"One could hardly ask for a better guide than Sebastian, who is, without exaggeration, the best machine learning educator currently in the field. On each page, Sebastian not only imparts his extensive knowledge but also shares the passion and curiosity that mark true expertise." "Chris Albon, Director of Machine Learning, The Wikimedia Foundation

"Sebastian Raschka's new book, Machine Learning Q and AI, is a one-stop shop for overviews of crucial AI topics beyond the core covered in most introductory courses"|If you have already stepped into the world of AI via deep neural networks, then this book will give you what you need to locate and understand the next level." "Ronald T. Kneusel, author of How AI Works



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Introduction

Thanks to rapid advancements in deep learning, we have seen a significant expansion of machine learning and AI in recent years.

This progress is exciting if we expect these advancements to create new industries, transform existing ones, and improve the quality of life for people around the world. On the other hand, the constant emergence of new techniques can make it challenging and time-consuming to keep abreast of the latest developments. Nonetheless, staying current is essential for professionals and organizations that use these technologies.

I wrote this book as a resource for readers and machine learning practitioners who want to advance their expertise in the field and learn about techniques that I consider useful and significant but that are often overlooked in traditional and introductory textbooks and classes. I hope you'll find this book a valuable resource for obtaining new insights and discovering new techniques you can implement in your work.

Tips: 本书会突出 **核心概念**，并且，会给出 **示例**，辅助理解。

Who Is This Book For?

Navigating the world of AI and machine learning literature can often feel like walking a tightrope, with most books positioned at either end: broad beginner's introductions or deeply mathematical treatises. This book illustrates and discusses important developments in these fields while staying approachable and not requiring an advanced math or coding background.

Tips: 本书，并不要求读者有高等数学知识、也无需编码背景。简单来说，普通的高中毕业，也可以流畅阅读。

This book is for people with some experience with machine learning who want to learn new concepts and techniques. It's ideal for those who have taken a beginner course in machine learning or deep learning or have read an equivalent

introductory book on the topic. (Throughout this book, I will use *machine learning* as an umbrella term for machine learning, deep learning, and AI.)



本书中，会使用 **机器学习** 作为 **统称**，包括机器学习、深度学习、AI。

What Will You Get Out of This Book?

This book adopts a unique Q&A style, where each brief chapter is structured around a central question related to fundamental concepts in machine learning, deep learning, and AI. Every question is followed by an explanation, with several illustrations and figures, as well as exercises to test your understanding. Many chapters also include references for further reading. These bite-sized nuggets of information provide an enjoyable jumping-off point on your journey from machine learning beginner to expert.

The book covers a wide range of topics. It includes new insights about established architectures, such as convolutional networks, that allow you to utilize these technologies more effectively. It also discusses more advanced techniques, such as the inner workings of large language models (LLMs) and vision transformers. Even experienced machine learning researchers and practitioners will encounter something new to add to their arsenal of techniques.

Tips: 本书，会介绍 **AI 领域** 的典型概念、知识，但不是数学或编码书籍。阅读时，无需证明或编码、突出易读性。

While this book will expose you to new concepts and ideas, it's not a math or coding book. You won't need to solve any proofs or run any code while reading. In other words, this book is a perfect travel companion or something you can read on your favorite reading chair with your morning coffee or tea.

How to Read This Book

Each chapter of this book is designed to be self-contained, offering you the freedom to jump between topics as you wish. When a concept from one chapter is explained in more detail in another, I've included chapter references you can follow to fill in gaps in your understanding.

本书每个章节，都是独立的，你可以跳过一些章节，直接阅读你感兴趣的章节。



However, there's a strategic sequence to the chapters. For example, the early chapter on embeddings sets the stage for later discussions on self-supervised learning and few-shot learning. For the easiest reading experience and the most comprehensive grasp of the content, my recommendation is to approach the book from start to finish.

然而，本书的章节，是有 **顺序** 的，建议从前往后阅读；因为，把 **最通用的概念**，放在了最前章节。

Each chapter is accompanied by optional exercises for readers who want to test their understanding, with an answer key located at the end of the book. In addition, for any papers referenced in a chapter or further reading on that chapter's topic, you can find the complete citation information in that chapter's "References" section.

The book is structured into five main parts centered on the most important topics in machine learning and AI today.

整体内容，分为 5 个部分，都是 **AI 领域** 的 **最重要的主题**。

Tips: 下面**第一部分**，是 **神经网络** 和 **深度学习** 的 **通用概念**，包含嵌入、自监督学习、少样本学习、彩票假设、过拟合、多 GPU 训练范式等。

Part I: Neural Networks and Deep Learning covers questions about deep neural networks and deep learning that are not specific to a particular subdomain. For example, we discuss alternatives to supervised learning and techniques for reducing overfitting, which is a common problem when using machine learning models for real-world problems where data is limited.

Chapter [\[ch01\]](#): Embeddings, Latent Space, and Representations

Delves into the distinctions and similarities between embedding vectors, latent vectors, and representations. Elucidates how these concepts help encode information in the context of machine learning.



Chapter [\[ch02\]](#): Self-Supervised Learning

Focuses on self-supervised learning, a method that allows neural networks to utilize large, unlabeled datasets in a supervised manner.

Chapter [\[ch03\]](#): Few-Shot Learning

Introduces few-shot learning, a specialized supervised learning technique tailored for small training datasets.

Chapter [\[ch04\]](#): The Lottery Ticket Hypothesis

Explores the idea that randomly initialized neural networks contain smaller, efficient subnetworks.

Chapter [\[ch05\]](#): Reducing Overfitting with Data

Addresses the challenge of overfitting in machine learning, discussing strategies centered on data augmentation and the use of unlabeled data to reduce overfitting.

Chapter [\[ch06\]](#): Reducing Overfitting with Model Modifications

Extends the conversation on overfitting, focusing on model-related solutions like regularization, opting for simpler models, and ensemble techniques.

Chapter [\[ch07\]](#): Multi-GPU Training Paradigms

Explains various training paradigms for multi-GPU setups to accelerate model training, including data and model parallelism.

Chapter [\[ch08\]](#): The Success of Transformers

Explores the popular transformer architecture, highlighting features like attention mechanisms, parallelization ease, and high parameter counts.

Chapter [\[ch09\]](#): Generative AI Models

Provides a comprehensive overview of deep generative models, which are used to produce various media forms, including images, text, and audio. Discusses the strengths and weaknesses of each model type.

Chapter [\[ch10\]](#): Sources of Randomness

Addresses the various sources of randomness in the training of deep neural networks that may lead to inconsistent and non-reproducible results during both training and inference. While randomness can be accidental, it can also be intentionally introduced by design.

Tips: 下面第二部分，是 **计算机视觉** 的 **典型概念**，包含 卷积神经网络、视觉变换器。



Part II: Computer Vision focuses on topics mainly related to deep learning but specific to computer vision, many of which cover convolutional neural networks and vision transformers.

Chapter [\[ch11\]](#): Calculating the Number of Parameters

Explains the procedure for determining the parameters in a convolutional neural network, which is useful for gauging a model's storage and memory requirements.

Chapter [\[ch12\]](#): Fully Connected and Convolutional Layers

Illustrates the circumstances in which convolutional layers can seamlessly replace fully connected layers, which can be useful for hardware optimization or simplifying implementations.

Chapter [\[ch13\]](#): Large Training Sets for Vision Transformers

Probes the rationale behind vision transformers requiring more extensive training sets compared to conventional convolutional neural networks.

Tips: 下面第三部分，文本相关，是 **自然语言处理** 的 **典型概念**，包含 分布式假设、数据增强、自注意力、编码器-解码器式变换器、使用和微调预训练变换器、评估生成式大语言模型等。

Part III: Natural Language Processing covers topics around working with text, many of which are related to transformer architectures and self-attention.

Chapter [\[ch14\]](#): The Distributional Hypothesis

Delves into the distributional hypothesis, a linguistic theory suggesting that words appearing in the same contexts tend to possess similar meanings, which has useful implications for training machine learning models.

Chapter [\[ch15\]](#): Data Augmentation for Text

Highlights the significance of data augmentation for text, a technique used to artificially increase dataset sizes, which can help with improving model performance.



Chapter [\[ch16\]](#): Self-Attention

Introduces self-attention, a mechanism allowing each segment of a neural network's input to refer to other parts. Self-attention is a key mechanism in modern large language models.

Chapter [\[ch17\]](#): Encoder- and Decoder-Style Transformers

Describes the nuances of encoder and decoder transformer architectures and explains which type of architecture is most useful for each language processing task.

Chapter [\[ch18\]](#): Using and Fine-Tuning Pretrained Transformers

Explains different methods for fine-tuning pretrained large language models and discusses their strengths and weaknesses.

Chapter [\[ch19\]](#): Evaluating Generative Large Language Models

Lists prominent evaluation metrics for language models like Perplexity, BLEU, ROUGE, and BERTScore.

Tips: 下面第四部分，是 **生产** 和 **部署** 的 **典型概念**，包含 无状态和有状态训练、数据分布偏移等。

Part IV: Production and Deployment covers questions pertaining to practical scenarios, such as increasing inference speeds and various types of distribution shifts.

Chapter [\[ch20\]](#): Stateless and Stateful Training

Distinguishes between stateless and stateful training methodologies used in deploying models.

Chapter [\[ch21\]](#): Data-Centric AI

Explores data-centric AI, which prioritizes refining datasets to enhance model performance. This approach contrasts with the conventional model-centric approach, which emphasizes improving model architectures or methods.

Chapter [\[ch22\]](#): Speeding Up Inference

Introduces techniques to enhance the speed of model inference without tweaking the model's architecture or compromising accuracy.



Chapter [ch23]: Data Distribution Shifts

Post-deployment, AI models

may face discrepancies between training data and real-world data distributions, known as data distribution shifts. These shifts can deteriorate model performance. This chapter categorizes and elaborates on common shifts like covariate shift, concept drift, label shift, and domain shift.

Tips: 下面第五部分，是 **预测性能** 和 **模型评估** 的 **典型概念**，包含泊松回归、置信区间、置信区间与一致性预测、交叉验证、训练和测试集不一致、有限标签数据等。

Part V: Predictive Performance and Model Evaluation dives deeper into various aspects of squeezing out predictive performance, such as changing the loss function, setting up k -fold cross-validation, and dealing with limited labeled data.

Chapter [ch24]: Poisson and Ordinal Regression

Highlights the differences between Poisson and ordinal regression. Poisson regression is suitable for count data that follows a Poisson distribution, like the number of colds contracted on an airplane. In contrast, ordinal regression caters to ordered categorical data without assuming equidistant categories, such as disease severity.

Chapter [ch25]: Confidence Intervals

Delves into methods for constructing confidence intervals for machine learning classifiers. Reviews the purpose of confidence intervals, discusses how they estimate unknown population parameters, and introduces techniques such as normal approximation intervals, bootstrapping, and retraining with various random seeds.

Chapter [ch26]: Confidence Intervals vs. Conformal Predictions

Discusses the distinction between confidence intervals and conformal predictions and describes the latter as a tool for creating prediction intervals that cover actual outcomes with specific probability.

Chapter [ch27]: Proper Metrics

Focuses on the essential properties of a proper metric in mathematics and computer science. Examines whether commonly used loss functions in machine



learning, such as mean squared error and cross-entropy loss, satisfy these properties.

Chapter [ch28]: The k in k -Fold Cross-Validation

Explores the role of the k in k -fold cross-validation and provides insight into the advantages and disadvantages of selecting a large k .

Chapter [ch29]: Training and Test Set Discordance

Addresses the scenario where a model performs better on a test dataset than the training dataset. Offers strategies to discover and address discrepancies between training and test datasets, introducing the concept of adversarial validation.

Chapter [ch30]: Limited Labeled Data

Introduces various techniques to enhance model performance in situations where data is limited. Covers data labeling, bootstrapping, and paradigms such as transfer learning, active learning, and multimodal learning.

Online Resources

I've provided optional supplementary materials on GitHub with code examples for certain chapters to enhance your learning experience (see <https://github.com/rasbt/MachineLearning-QandAI-book>). These materials are designed as practical extensions and deep dives into topics covered in the book. You can use them alongside each chapter or explore them after reading to solidify and expand your knowledge.

Without further ado, let's dive in.



Chapter 1: Embeddings, Latent Space, and Representations

In deep learning, we often use the terms *embedding vectors*, *representations*, and *latent space*. What do these concepts have in common, and how do they differ?

While these three terms are often used interchangeably, we can make subtle distinctions between them:

- **Embedding vectors** are representations of input data where similar items are close to each other.
- **Latent vectors** are intermediate representations of input data.
- **Representations** are encoded versions of the original input.

The following sections explore the relationship between embeddings, latent vectors, and representations and how each functions to encode information in machine learning contexts.

Embeddings

Embedding vectors, or *embeddings* for short, encode relatively high-dimensional data into relatively low-dimensional vectors.

Tips: 嵌入向量，简称 **嵌入**，是输入数据的一种表示形式，相似的输入、对应的嵌入向量 **彼此接近**；通常，将高维数据，转换为低维嵌入向量。

We can apply embedding methods to create a continuous dense (non-sparse) vector from a (sparse) **one-hot** encoding.

One-hot encoding is a method used to represent categorical data as binary vectors, where each category is mapped to a vector containing 1 in the position corresponding to the category's index, and 0 in all other positions.



This ensures that the categorical values are represented in a way that certain machine learning algorithms can process. For example, if we have a categorical variable Color with three categories, Red, Green, and Blue, the one-hot encoding would represent Red as [1, 0, 0], Green as [0, 1, 0], and Blue as [0, 0, 1]. These one-hot encoded categorical variables can then be mapped into continuous embedding vectors by utilizing the learned weight matrix of an embedding layer or module.

独热编码：**one-hot** 编码，是一种将分类数据转换为 **二进制向量** 的方法，其中每个类别映射到包含 1 的向量，在对应类别的索引位置为1，其他位置为0。

We can also use embedding methods for dense data such as images. For example, the last layers of a convolutional neural network may yield embedding vectors, as illustrated in Figure 1.1 .

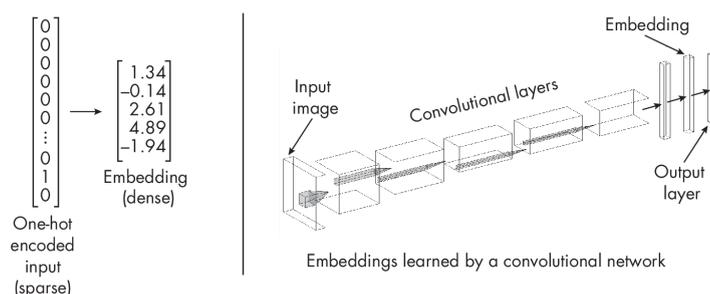


Figure 1.1

To be technically correct, all intermediate layer outputs of a neural network could yield embedding vectors. Depending on the training objective, the output layer may also produce useful embedding vectors. For the sake of simplicity, the convolutional neural network in Figure 1.1

Embeddings can have higher or lower numbers of dimensions than the original input. For instance, using embeddings methods for extreme expression, we can encode data into two-dimensional dense and continuous representations for visualization purposes and clustering analysis, as illustrated in Figure 1.2.

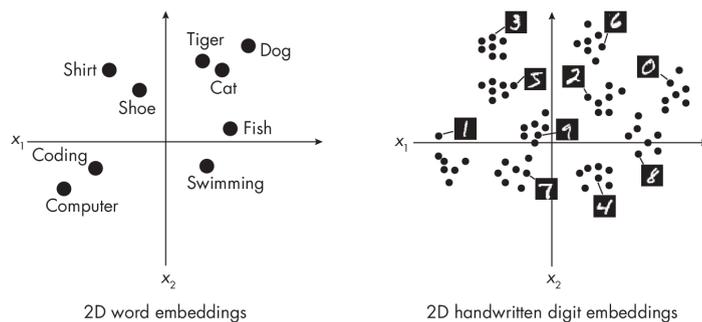


Figure 1.2

A **fundamental property** of embeddings is that they encode *distance* or *similarity*. This means that embeddings capture the semantics of the data such that similar inputs are close in the embeddings space.

Tips: 嵌入向量，具有一个重要的性质，即编码 **距离相近** 或 **相似性**。这意味着嵌入向量能够捕捉数据的语义，使得相似的输入在嵌入空间中彼此接近。这也称为 **结构保持** structure-preserving 特性。

For readers interested in a more formal explanation using mathematical terminology, an embedding is an **injective** and **structure-preserving** map between an input space X and the embedding space Y . This implies that similar inputs will be located at points in close proximity within the embedding space, which can be seen as the "structure-preserving" characteristic of the embedding.

Tips: 嵌入向量，是输入空间 X 和嵌入空间 Y 之间的一个**单向**和**结构保持**映射。这意味着相似的输入，在嵌入空间中彼此接近，这就是 **结构保持** 特性。

Latent Space

Latent space is typically used synonymously with *embedding space*, the space into which embedding vectors are mapped.

Tips: 隐空间 \ 潜空间，通常与**嵌入空间**同义，是 **嵌入向量** 映射到的空间。



Similar items can appear close in the latent space; however, this is not a strict requirement. More loosely, we can think of the latent space as any feature space that contains features, often compressed versions of the original input features. These latent space features can be learned by a neural network, such as an **autoencoder** that reconstructs input images, as shown in Figure 1.3.

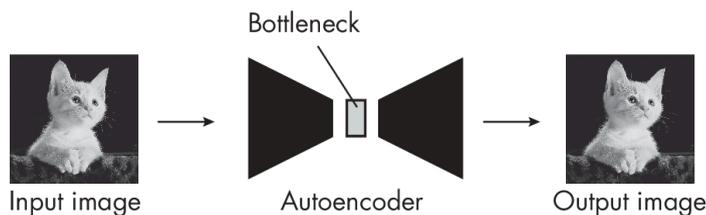


Figure 1.3

The **bottleneck** in Figure 1.3 represents a small, intermediate neural network layer that encodes or maps the input image into a lower-dimensional representation. We can think of the target space of this mapping as a latent space. The training objective of the autoencoder is to reconstruct the input image, that is, to minimize the distance between the input and output images. In order to optimize the training objective, the autoencoder may learn to place the encoded features of similar inputs (for example, pictures of cats) close to each other in the latent space, thus creating useful embedding vectors where similar inputs are close in the embedding (latent) space.

Representation

A *representation* is an encoded, typically intermediate form of an input. For instance, an embedding vector or vector in the latent space is a representation of the input, as previously discussed. However, representations can also be produced by simpler procedures. For example, one-hot encoded vectors are considered representations of an input.

The key idea is that the representation captures some essential features or characteristics of the original data to make it useful for further analysis or processing.

Tips: 表示/表征 representation，是输入的一种编码形式，通常是中间形式。关键点是，它能够捕捉输入的一些 **本质特征** 或 **特性**，可用于后续分析。



Exercises

1-1. Suppose we're training a convolutional network with five convolutional layers followed by three fully connected (FC) layers, similar to AlexNet

(<https://en.wikipedia.org/wiki/AlexNet>), as illustrated in Figure 1.4.

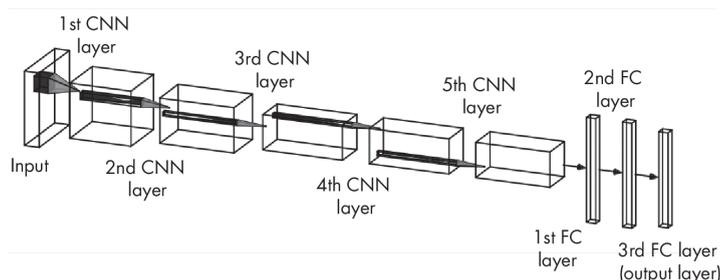
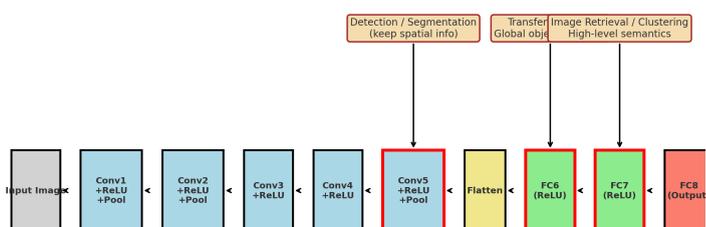


Figure 1.4

We can think of these fully connected layers as two hidden layers and an output layer in a multilayer perceptron. Which of the neural network layers can be utilized to produce useful embeddings? Interested readers can find more details about the AlexNet architecture and implementation in the original publication by Alex Krizhevsky, Ilya Sutskever, and Geoffrey Hinton.

AlexNet Embedding Extraction Points and Their Applications



Answer 1-1

AlexNet 的完整流程：

- 输入图像 → 卷积层堆叠 (Conv1–Conv5) → 展平 → 全连接层 (FC6–FC8)
- 标红的 Conv5、FC6、FC7 是常用来提取有用 embedding 特征的层。
 - Conv5 → 检测 / 分割（保留空间位置信息）
 - FC6 → 迁移学习（整体结构特征）
 - FC7 → 图像检索 / 聚类（高级语义表示）

Answer:



asking: in an AlexNet-style convolutional neural network (five conv layers + three >FC layers), which layers can provide useful embeddings that capture semantic >information about the input.

Let's break it down:

1. Convolutional Layers (early to mid layers)

- **Early conv layers (conv1–conv2):** These capture **low-level features** like edges, corners, textures, and simple >shapes. → Not very useful as "semantic embeddings," but good for transfer to tasks >requiring fine textures (e.g., medical images, style).
- **Mid-depth conv layers (conv3–conv5):** These capture **mid- to high-level features** (object parts, motifs, hierarchical >compositions). → Often very useful as **general embeddings** for vision tasks like detection, >segmentation, or retrieval. (In fact, features from conv5 of AlexNet were widely used in early transfer >learning.)

2. Fully Connected Layers

- **First two FC layers (fc6, fc7):** These act like dense "semantic bottlenecks."
- **fc6** embeddings capture global object structure.
- **fc7** embeddings are particularly popular for transfer learning and >representation learning because they're **highly discriminative yet not tied to the >final class labels**. → Both are excellent choices for embeddings.
- **Last FC layer (fc8, the classifier):** This outputs class scores (e.g., 1000-dim for ImageNet). → Too task-specific (logits for specific labels). Not generally useful as >embeddings unless you remove the softmax and use the logits for re-scoring.

3. Best Practices

- If you want **general embeddings** (for clustering, retrieval, transfer learning):



- Use `conv5` (after flatten + pooling) or `fc6` / `fc7` .
- `fc7` is most common, since it captures high-level semantics.
- If you want **task-specific representations**:
- Use `fc8` logits.
- If you want **visual features for downstream CV models (e.g., detection, >segmentation)**:
- Use convolutional feature maps (conv3–conv5), since they preserve spatial >structure.

4. Summary:

Useful embeddings can be taken from **the last convolutional layer (conv5) or the >penultimate fully connected layers (fc6, fc7)**. The final output layer (fc8) is >usually not used for embeddings since it's tied to a specific classification task.

1-2. Name some types of input representations that are not embeddings.

References

- The original paper describing the AlexNet architecture and implementation: Alex Krizhevsky, Ilya Sutskever, and Geoffrey Hinton, "ImageNet Classification with Deep Convolutional Neural Networks" (2012), <https://papers.nips.cc/paper/4824-imagenet-classification-with-deep-convolutional-neural-networks>.

Chapter 2: Self-Supervised Learning



What is self-supervised learning, when is it useful, and what are the main approaches to implementing it?

Self-supervised learning is a pretraining procedure that lets neural networks leverage large, unlabeled datasets in a supervised fashion. This chapter compares self-supervised learning to transfer learning, a related method for pretraining neural networks, and discusses the practical applications of self-supervised learning. Finally, it outlines the main categories of self-supervised learning.

Tips: 自监督学习，是一种预训练方法，让神经网络利用 **无标签** 的大数据集，进行 **监督学习**。其实，使用 **无标签** 数据，自动构造了 **伪标签**（例如：遮挡部分内容、预测缺失内容），也是一种 **监督学习**。

Self-Supervised Learning vs. Transfer Learning

Self-supervised learning is related to **transfer learning**, a technique in which a model pretrained on one task is reused as the starting point for a model on a second task. For example, suppose we are interested in training an image classifier to classify bird species. In transfer learning, we would pretrain a convolutional neural network on the ImageNet dataset, a large, labeled image dataset with many different categories, including various objects and animals. After pretraining on the general ImageNet dataset, we would take that pretrained model and train it on the smaller, more specific target dataset that contains the bird species of interest. (Often, we just have to change the class-specific output layer, but we can otherwise adopt the pretrained network as is.)

Figure [2.1](#) illustrates the process of transfer learning.

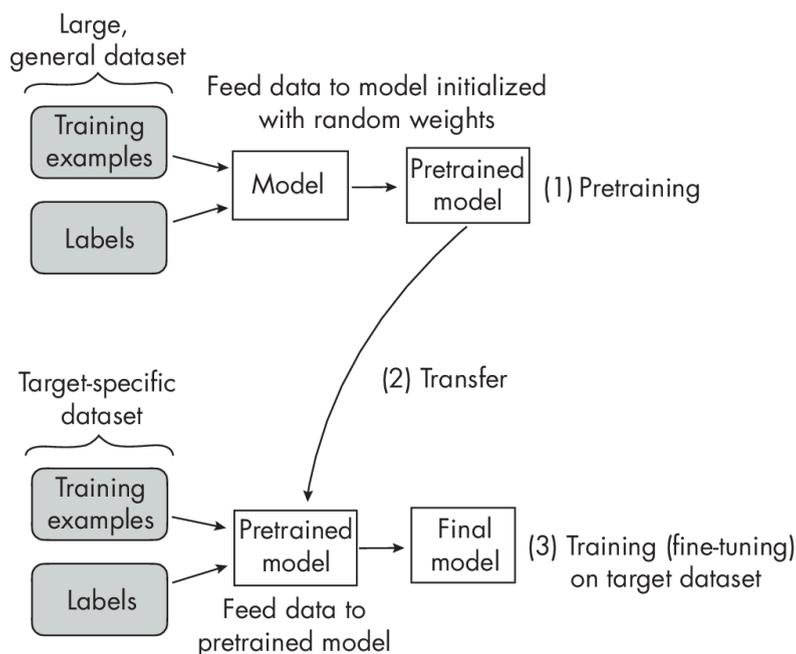


Figure 2.1

Tips: 自监督学习，与 迁移学习 是相关的。

- 相同点在于，都是使用 **预训练** 的模型，然后进行 **微调**。
- 差异在于，迁移学习是使用 **有标签** 的数据，而自监督学习使用 **无标签** 的数据。

Self-supervised learning is an alternative approach to transfer learning in which the model is pretrained not on labeled data but on *unlabeled* data. We consider an unlabeled dataset for which we do not have label information, and then we find a way to obtain labels from the dataset's structure to formulate a prediction task for the neural network, as illustrated in Figure 2.2. These self-supervised training tasks are also called *pretext tasks*.

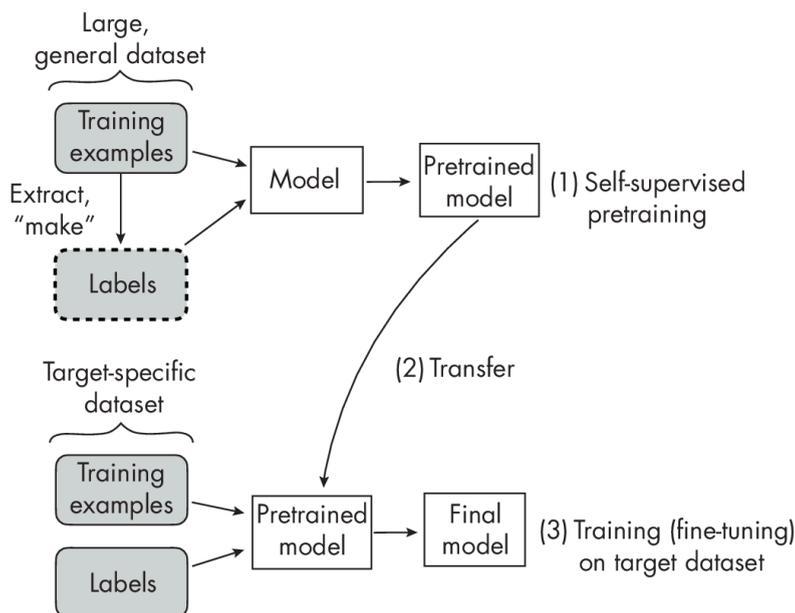


Figure 2.2

The main difference between transfer learning and self-supervised learning lies in how we obtain the labels during step 1 in Figures 2.1 and 2.2. In transfer learning, we assume that the labels are provided along with the dataset; they are typically created by human labelers. In self-supervised learning, the labels can be directly derived from the training examples.

Tips: 自监督学习中，数据集的标签，可以 **直接** 从训练样本中 **推导** 出来。

A self-supervised learning task could be a missing-word prediction in a natural language processing context. For example, given the sentence "It is beautiful and sunny outside," we can mask out the word *sunny*, feed the network the input "It is beautiful and [MASK] outside," and have the network predict the missing word in the "[MASK]" location. Similarly, we could remove image patches in a computer vision context and have the neural network fill in the blanks. These are just two examples of self-supervised learning tasks; many more methods and paradigms for this type of learning exist.

In sum, we can think of self-supervised learning on the pretext task as *representation learning*. We can take the pretrained model to fine-tune it on the target task (also known as the *downstream* task).

Leveraging Unlabeled Data



Large neural network architectures require large amounts of labeled data to perform and generalize well. However, for many problem areas, we don't have access to large labeled datasets. With self-supervised learning, we can leverage unlabeled data. Hence, self-supervised learning is likely to be useful when working with large neural networks and with a limited quantity of labeled training data.

Transformer-based architectures that form the basis of LLMs and vision transformers are known to require self-supervised learning for pretraining to perform well.

For small neural network models such as multilayer perceptrons with two or three layers, self-supervised learning is typically considered neither useful nor necessary.

Tips: 对于 **小型** 的神经网络模型，如具有两到三层的 **多层感知器**，**自监督学习** 在这种情况下 **不实用也 不必要**。

Self-supervised learning likewise isn't useful in traditional machine learning with nonparametric models such as tree-based random forests or gradient boosting. Conventional tree-based methods do not have a fixed parameter structure (in contrast to the weight matrices, for example). Thus, conventional tree-based methods are not capable of transfer learning and are incompatible with self-supervised learning.

Tips: 对于 **非参数模型**，如基于树的随机森林或梯度提升，**自监督学习** 通常不适用。

传统的基于树的方法没有固定的参数结构（与权重矩阵相比），因此传统的基于树的方法无法进行迁移学习，也不兼容自监督学习。

FIXME 没理解???

Self-Prediction and Contrastive Self-Supervised Learning



There are two main categories of self-supervised learning: **self-prediction** and **contrastive self-supervised** learning. In *self-prediction*, illustrated in Figure 2.3, we typically change or hide parts of the input and train the model to reconstruct the original inputs, such as by using a perturbation mask that obfuscates certain pixels in an image.

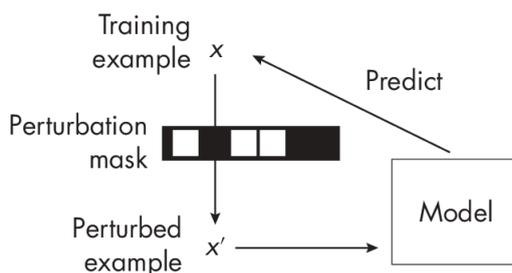


Figure 2.3

A classic example is a denoising autoencoder that learns to remove noise from an input image. Alternatively, consider a masked autoencoder that reconstructs the missing parts of an image, as shown in Figure 2.4.

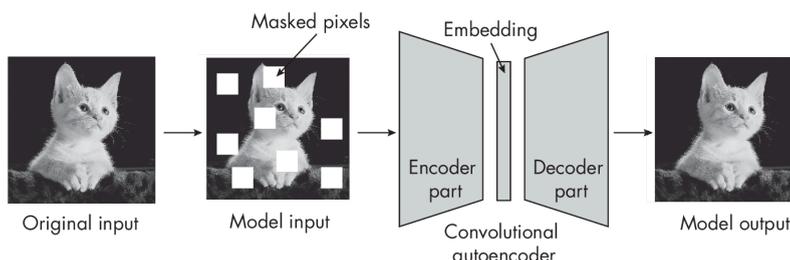


Figure 2.4

Missing (**masked**) input self-prediction methods are also commonly used in natural language processing contexts. Many generative LLMs, such as GPT, are trained on a next-word prediction pretext task (GPT will be discussed at greater length in Chapters [\[ch14\]](#) and [\[ch17\]](#). Here, we feed the network text fragments, where it has to predict the next word in the sequence (as we'll discuss further in Chapter [\[ch17\]](#)).

In *contrastive self-supervised learning*, we train the neural network to learn an embedding space where similar inputs are close to each other and dissimilar inputs are far apart. In other words, we train the network to produce embeddings that minimize the distance between similar training inputs and maximize the distance between dissimilar training examples.



Let's discuss contrastive learning using concrete example inputs. Suppose we have a dataset consisting of random animal images. First, we draw a random image of a cat (the network does not know the label, because we assume that the dataset is unlabeled). We then augment, corrupt, or perturb this cat image, such as by adding a random noise layer and cropping it differently, as shown in Figure 2.5.

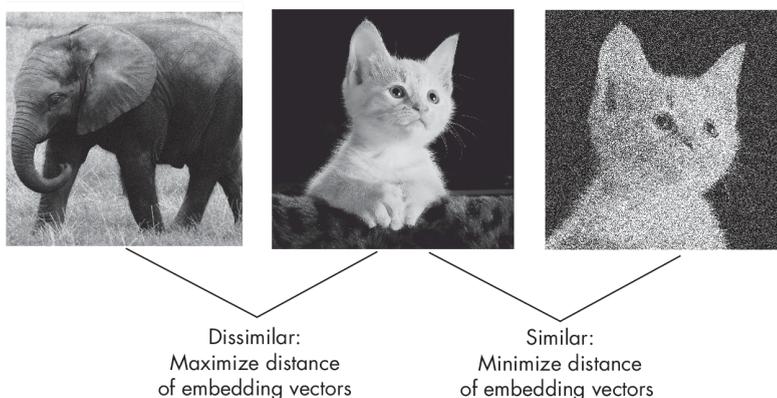


Figure 2.5

The perturbed cat image in this figure still shows the same cat, so we want the network to produce a similar embedding vector. We also consider a random image drawn from the training set (for example, an elephant, but again, the network doesn't know the label).

For the cat-elephant pair, we want the network to produce dissimilar embeddings. This way, we implicitly force the network to capture the image's core content while being somewhat agnostic to small differences and noise. For example, the simplest form of a contrastive loss is the L_2 -norm (Euclidean distance) between the embeddings produced by model $M(\cdot)$. Let's say we update the model weights to decrease the distance $\|M(cat) - M(cat')\|_2$ and increase the distance $\|M(cat) - M(elephant)\|_2$.

Figure 2.6 summarizes the central concept behind **contrastive learning** for the perturbed image scenario. The model is shown twice, which is known as a *siamese network* setup. Essentially, the same model is utilized in two instances: first, to generate the embedding for the original training example, and second, to produce the embedding for the perturbed version of the sample.

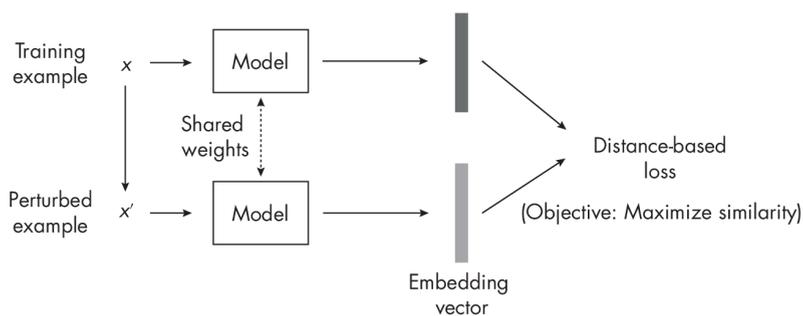


Figure 2.6

This example outlines the main idea behind contrastive learning, but many subvariants exist. Broadly, we can categorize these into *sample* contrastive and *dimension* contrastive methods. The elephant-cat example in Figure 2.6 illustrates a sample contrastive method, where we focus on learning embeddings to minimize and maximize distances between training pairs. In *dimension*-contrastive approaches, on the other hand, we focus on making only certain variables in the embedding representations of similar training pairs appear close to each other while maximizing the distance of others.

Tips: 对比学习，可以分为 **样本对比** 和 **维度对比** 两种方法。

- 样本对比，关注于学习嵌入，以最小化/最大化 **训练对** 之间的距离。
- 维度对比，关注于使相似 **训练对** 中的**某些变量**接近，同时最大化其他变量的距离。

Exercises

2-1. How could we apply self-supervised learning to video data?

2-2. Can self-supervised learning be used for tabular data represented as rows and columns? If so, how could we approach this?

Talk

Talk 1: 自监督学习，使用了无标签数据，进行预训练，本质也是一种有监督训练，理解对吗？



你的理解非常接近，但里面有一点细微差别，我帮你梳理下：

1. 传统监督学习

- 有人工标注的标签 y 。
- 目标函数直接基于 (x, y) 优化（例如图像分类：输入图片，标签是「猫/狗」）。

2. 自监督学习 (Self-Supervised Learning, SSL)

- 使用无标签数据（比如大量文本、图片、语音）。
- 但它会 自动生成“伪标签” (pretext task)，来构造监督信号。例子：
 - NLP 里 **掩码预测 (masked language modeling)**：给模型输入“我去 [MASK] 吃饭”，伪标签就> 是被遮住的词。
 - CV 里 **对比学习**：从同一张图生成两个不同增强视图，伪标签就是“这俩是同一个对象”。

换句话说，自监督 \neq 无监督。它其实是监督学习的一种特例，只不过监督信号不是人手标注，而是 从数据本身挖掘出来的监督信息。

3. 本质理解

- 是监督学习：因为它还是通过“预测某个目标”来训练，损失函数需要“标签”（哪怕是自生成的）。
- 区别在于标签的来源：
 - 监督学习 \rightarrow 外部人工提供。
 - 自监督学习 \rightarrow 数据内部挖掘。

4. 结论

所以你的说法「自监督学习，本质也是一种有监督训练」是正确的，只是要更精确地说：

- 它是一种“利用数据自身生成监督信号的监督学习”。



References

- For more on the ImageNet dataset: <https://en.wikipedia.org/wiki/ImageNet>.
 - An example of a contrastive self-supervised learning method: Ting Chen et al., "A Simple Framework for Contrastive Learning of Visual Representations"? (2020), <https://arxiv.org/abs/2002.05709>.
 - An example of a dimension-contrastive method: Adrien Bardes, Jean Ponce, and Yann LeCun, "VICRegL: Self-Supervised Learning of Local Visual Features"? (2022), <https://arxiv.org/abs/2210.01571>.
 - If you plan to employ self-supervised learning in practice: Randall Balestriero et al., "A Cookbook of Self-Supervised Learning"? (2023), <https://arxiv.org/abs/2304.12210>.
 - A paper proposing a method of transfer learning and self-supervised learning for relatively small multilayer perceptrons on tabular datasets: Dara Bahri et al., "SCARF: Self-Supervised Contrastive Learning Using Random Feature Corruption"? (2021), <https://arxiv.org/abs/2106.15147>.
 - A second paper proposing such a method: Roman Levin et al., "Transfer Learning with Deep Tabular Models"? (2022), <https://arxiv.org/abs/2206.15306>.
-



Chapter 3: Few-Shot Learning

What is few-shot learning? How does it differ from the conventional training procedure for supervised learning?

Few-shot learning is a type of supervised learning for small training sets with a very small example-to-class ratio. In regular supervised learning, we train models by iterating over a training set where the model always sees a **fixed set** of classes. In few-shot learning, we are working on a **support set** from which we create multiple training tasks to assemble training episodes, where each training task consists of different classes.

Tips:

- 小样本学习，关注于学习模型，以适应新的任务。
- 在传统的监督学习中，我们通过迭代训练集来训练模型，模型总是看到固定的类集。
- 在小样本学习中，我们从一个 **支持集** 开始，创建 **多个训练任务** 来组装训练集，每个训练任务包含不同的分类。

Datasets and Terminology

In supervised learning, we fit a model on a **training dataset** and evaluate it on a **test dataset**. The training set typically contains a relatively large number of examples per class. For example, in a supervised learning context, the Iris dataset, which has 50 examples per class, is considered a tiny dataset. For deep learning models, on the other hand, even a dataset like MNIST that has 5,000 training examples per class is considered very small.

In **few-shot learning**, the number of examples per class is much smaller. When specifying the few-shot learning task, we typically use the term *N-way K-shot*, where

- N stands for the number of classes
- and K stands for the number of examples per class.

The most common values are $K = 1$ or $K = 5$. For instance, in a 5-way 1-shot problem, there are five classes with only one example each. Figure [3.1](#) depicts a 3-



way 1-shot setting to illustrate the concept with a smaller example.

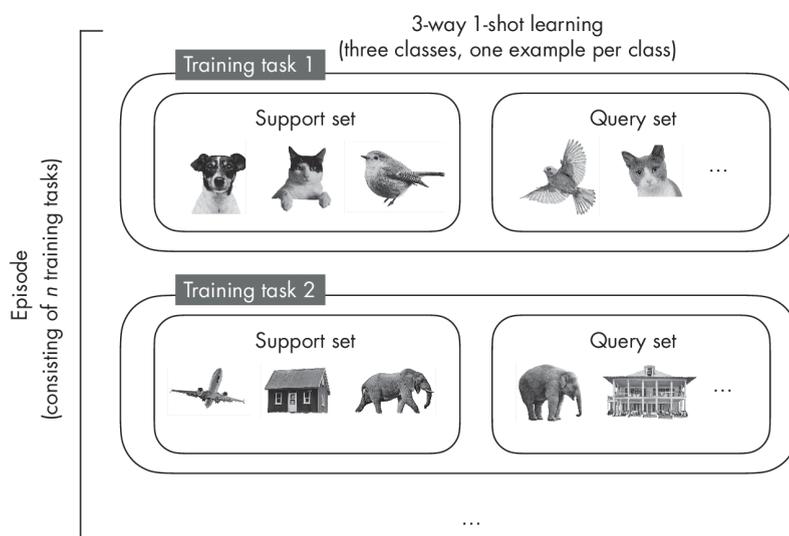


Figure 3.1

Rather than fitting the model to the training dataset, we can think of **few-shot learning** as "learning to learn."? In contrast to supervised learning, **few-shot learning** uses not a training dataset but a so-called **support set**, from which we sample training tasks that mimic the use-case scenario during prediction. With each training task comes a query image to be classified. The model is trained on several training tasks from the support set; this is called an **episode**.

Tips: 小样本学习，可以看作是 **学习如何学习**。

与传统的监督学习不同，小样本学习不使用训练集，而是使用所谓的 **支持集**，从中采样训练任务，以模仿预测时的使用场景。

每个训练任务都有一个查询图像需要分类。

模型在支持集的多个训练任务上进行训练；这称为 **一个训练轮次**。

FIXME??? 不理解

Next, during testing, the model receives a new task with classes different from those seen during training. The classes encountered in training are also called **base classes**, and the support set during training is also often called the **base set**. Again, the task is to classify the query images. Test tasks are similar to training tasks, except that none of the classes during testing overlap with those encountered during training, as illustrated in Figure 3.2.

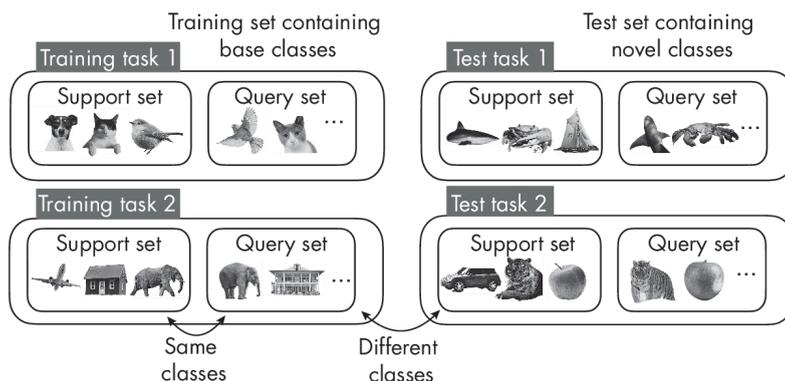


Figure 3.2

As Figure 3.2 shows, the support and query sets contain different images from the same class during training. The same is true during testing. However, notice that the classes in the support and query sets differ from the support and query sets encountered during training.

There are many different types of few-shot learning. In the most common, *meta-learning*, training is essentially about updating the model's parameters such that it can *adapt* well to a new task. On a high level, one few-shot learning strategy is to learn a model that produces embeddings where we can find the target class via a nearest-neighbor search among the images in the support set. Figure 3.3 illustrates this approach.

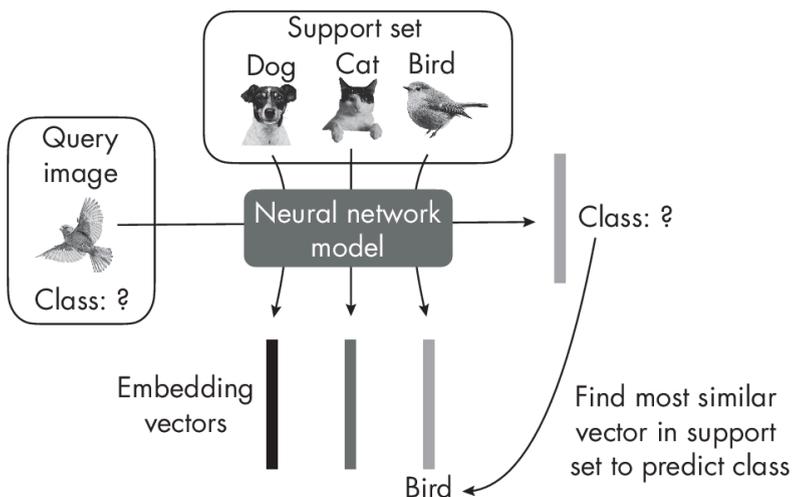


Figure 3.3

The model learns how to produce good embeddings from the support set to classify the query image based on finding the most similar embedding vector.

Exercises



3-1. MNIST (https://en.wikipedia.org/wiki/MNIST_database) is a classic and popular machine learning dataset consisting of 50,000 handwritten digits from 10 classes corresponding to the digits 0 to 9. How can we partition the MNIST dataset for a one-shot classification context?

3-2. What are some real-world applications or use cases for few-shot learning?



Chapter 4: The Lottery Ticket Hypothesis

What is the lottery ticket hypothesis, and, if it holds true, how is it useful in practice?

The lottery ticket hypothesis is a concept in neural network training that posits that within a randomly initialized neural network, there exists a **subnetwork** (or **winning ticket** ?) that can, when trained separately, achieve the same accuracy on a test set as the full network after being trained for the same number of steps. This idea was first proposed by Jonathan Frankle and Michael Carbin in 2018.

Tips: 彩票假设 **lottery ticket hypothesis**，是神经网络训练中，一个重要的概念。它指出，在随机初始化的神经网络中，存在一个 **子网络**（或 **彩票**），当单独训练时，可以达到与完整网络相同的准确率。

This chapter illustrates the lottery hypothesis step by step, then goes over *weight pruning*, one of the key techniques to create smaller subnetworks as part of the lottery hypothesis methodology. Lastly, it discusses the practical implications and limitations of the hypothesis.

Tips: 本章将展示 **彩票假设** 的训练过程，然后介绍 **权重剪枝**，这是 **彩票假设** 方法论中，创建较小子网络的关键技术。最后，讨论 **彩票假设** 的实际应用和局限性。

The Lottery Ticket Training Procedure

Figure [4.1](#) illustrates the training procedure for the lottery ticket hypothesis in four steps, which we'll discuss one by one to help clarify the concept.

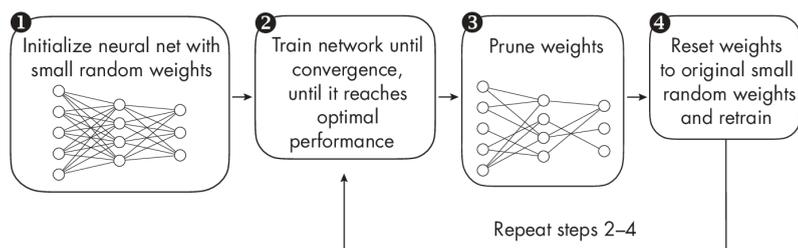


Figure 4.1

In Figure 4.1, we start with a large **neural network** that we train until convergence, meaning we put in our best efforts to make it perform as well as possible on a target dataset (for example, minimizing training loss and maximizing classification accuracy). This large neural network is initialized as usual using small random weights.

Next, as shown in Figure 4.1, we **prune** the neural network's **weight** parameters, removing them from the network. We can do this by setting the weights to zero to create sparse weight matrices. Here, we can either prune individual weights, known as *unstructured* pruning, or prune larger "chunks" from the network, such as entire convolutional filter channels. This is known as *structured* pruning.

Tips: 剪枝时，有两种方式，一种是 **unstructured pruning**，一种是 **structured pruning**。他们的差异是，**unstructured pruning** 是逐个剪枝，而 **structured pruning** 是剪枝整个 **卷积核**。

The original lottery hypothesis approach follows a concept known as *iterative magnitude pruning*, where the weights with the lowest magnitudes are removed in an iterative fashion. (We will revisit this concept in Chapter [ch06] when discussing techniques to reduce overfitting.)

Tips: 迭代剪枝 **iterative magnitude pruning**。

After the pruning step, we reset the weights to the original small random values used in step 1 in Figure 4.1 and train the pruned network. It's worth emphasizing that we do not reinitialize the pruned network with any small random weights (as is typical for iterative magnitude pruning), and instead we reuse the weights from step 1.

Tips: 剪枝后，我们重置权重为原始小随机值，并训练剪枝后的网络。??? 没理解 FIXME



We then repeat the pruning steps 2 through 4 until we reach the desired network size. For example, in the original lottery ticket hypothesis paper, the authors successfully reduced the network to 10 percent of its original size without sacrificing classification accuracy. As a nice bonus, the pruned (sparse) network, referred to as the *winning ticket*, even demonstrated improved generalization performance compared to the original (large and dense) network.

Practical Implications and Limitations

If it's possible to identify smaller subnetworks that have the same predictive performance as their up-to-10-times-larger counterparts, this can have significant implications for both neural training and inference. Given the ever-growing size of modern neural network architectures, this can help cut training costs and infrastructure.

Tips: 如果可以识别出与完整网络具有相同预测性能的较小子网络，这对于神经网络的 **训练** 和 **推理** 都有显著的影响，可以显著 **降低训练成本** 和 **基础设施成本**。

Sound too good to be true? Maybe. If winning tickets can be identified efficiently, this would be very useful in practice. However, at the time of writing, there is no way to find the winning tickets without training the original network. Including the pruning steps would make this even more expensive than a regular training procedure. Moreover, after the publication of the original paper, researchers found that the original weight initialization may not work to find winning tickets for larger-scale networks, and additional experimentation with the initial weights of the pruned networks is required.

The good news is that winning tickets do exist. Even if it's currently not possible to identify them without training their larger neural network counterparts, they can be used for more efficient inference after training.



Exercises

4-1. Suppose we're trying out the lottery ticket hypothesis approach and find that the performance of the subnetwork is not very good (compared to the original network). What next steps might we try?

4-2. The simplicity and efficiency of the rectified linear unit (ReLU) activation function have made it one of the most popular activation functions in neural network training, particularly in deep learning, where it helps to mitigate problems like the vanishing gradient. The ReLU activation function is defined by the mathematical expression $\max(0, x)$. This means that if the input x is positive, the function returns x , but if the input is negative or 0, the function returns 0. How is the lottery ticket hypothesis related to training a neural network with ReLU activation functions?

References

- The paper proposing the lottery ticket hypothesis: Jonathan Frankle and Michael Carbin, "The Lottery Ticket Hypothesis: Finding Sparse, Trainable Neural Networks"? (2018), <https://arxiv.org/abs/1803.03635>.
- The paper proposing structured pruning for removing larger parts, such as entire convolutional filters, from a network: Hao Li et al., "Pruning Filters for Efficient ConvNets"? (2016), <https://arxiv.org/abs/1608.08710>.
- Follow-up work on the lottery hypothesis, showing that the original weight initialization may not work to find winning tickets for larger-scale networks, and additional experimentation with the initial weights of the pruned networks is required: Jonathan Frankle et al., "Linear Mode Connectivity and the Lottery Ticket Hypothesis"? (2019), <https://arxiv.org/abs/1912.05671>.
- An improved lottery ticket hypothesis algorithm that finds smaller networks that match the performance of a larger network exactly: Vivek Ramanujan et al., "What's Hidden in a Randomly Weighted Neural Network"? (2020), <https://arxiv.org/abs/1911.13299>.



Chapter 5: Reducing Overfitting with Data

Suppose we train a neural network classifier in a supervised fashion and notice that it suffers from overfitting. What are some of the common ways to reduce overfitting in neural networks through the use of altered or additional data?

Overfitting, a common problem in machine learning, occurs when a model fits the training data too closely, learning its noise and outliers rather than the underlying pattern. As a result, the model performs well on the training data but poorly on unseen or test data. While it is ideal to prevent overfitting, it's often not possible to completely eliminate it. Instead, we aim to reduce or minimize overfitting as much as possible.

The most successful techniques for reducing overfitting revolve around collecting more high-quality labeled data. However, if collecting more labeled data is not feasible, we can **augment** the existing data or leverage unlabeled data for **pretraining**.

Tips: 减少 **过拟合**，最有效的技术是收集更多 **高质量的标签数据**；此外，还可以使用 **数据增强** 和 **预训练** 等技术。

Common Methods

This chapter summarizes the most prominent examples of dataset-related techniques that have stood the test of time, grouping them into the following categories: **collecting more data**, **data augmentation**, and **pretraining**.

Collecting More Data

One of the best ways to reduce overfitting is to collect more (good-quality) data. We can plot learning curves to find out whether a given model would benefit from more data. To construct a learning curve, we train the model to different training set sizes (10 percent, 20 percent, and so on) and evaluate the trained model on the same fixed-size validation or test set. As shown in Figure [5.1](#), the validation



accuracy increases as the training set sizes increase. This indicates that we can improve the model's performance by collecting more data.

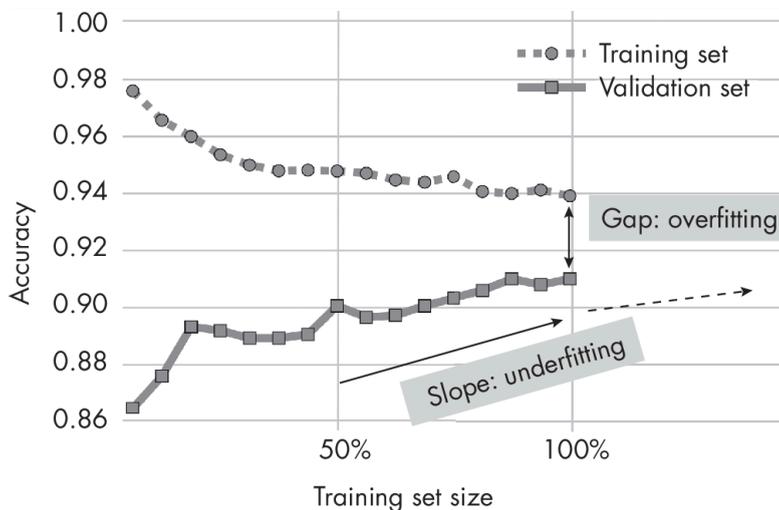


Figure 5.1

The gap between training and validation performance indicates the degree of overfitting--the more extensive the gap, the more overfitting occurs. Conversely, the slope indicating an improvement in the validation performance suggests the model is **underfitting** and can benefit from more data. Typically, **additional data** can decrease both **underfitting** and **overfitting**.

Data Augmentation

Data augmentation refers to generating new data records or features based on existing data. It allows for the expansion of a dataset without additional data collection.

Tips: **数据增强 data augmentation**，是一种常用的技术，用于增加数据集的大小和多样性。它通过 **生成新的数据**，来扩展数据集，而不需要额外的数据收集。

Data augmentation allows us to create different versions of the original input data, which can improve the model's generalization performance. Why? Augmented data can help the model improve its ability to generalize, since it makes it harder to memorize spurious information via training examples or features--or, in the case of image data, exact pixel values for specific pixel locations.



Tips: 数据增强 **data augmentation**，可以提高模型的泛化性能。为什么？因为增强后的数据，使模型忽略 **虚假信息**，典型场景：在 **图像数据** 中，会弱化 **特定像素** 的 **像素值**。

Figure 5.2 highlights common image data augmentation techniques, including **increasing brightness**, **flipping**, and **cropping**.

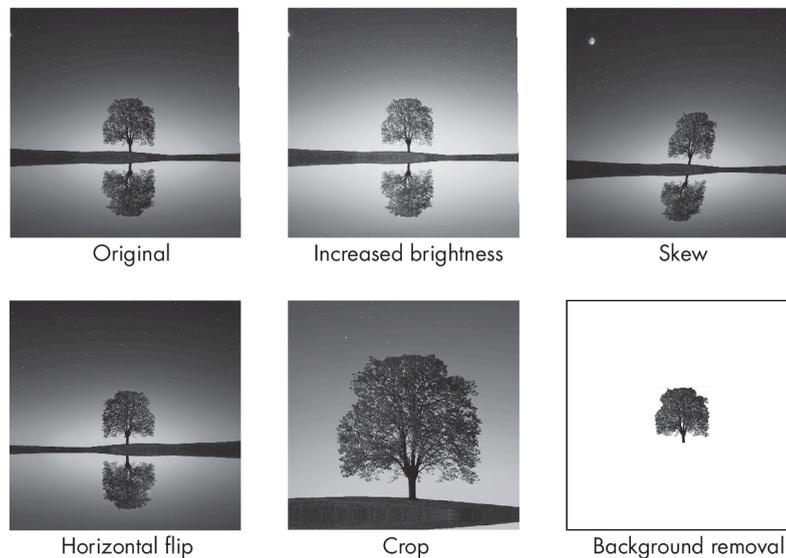


Figure 5.2

Data augmentation is usually standard for image data (see Figure 5.2) and text data (discussed further in Chapter [ch15]), but data augmentation methods for tabular data also exist.

Instead of collecting more data or augmenting existing data, it is also possible to generate new, **synthetic data**. While more common for image data and text, generating synthetic data is also possible for tabular datasets.

Tips: 数据增强 **data augmentation**，是图像数据和文本数据的标准技术。除了 **数据增强**，还可以 **生成合成数据**。

Pretraining

As discussed in Chapter [ch02], self-supervised learning lets us leverage large, unlabeled datasets to pretrain neural networks. This can also help reduce overfitting on the smaller target datasets.



As an alternative to **self-supervised learning** , **traditional transfer learning** on large labeled datasets is also an option. Transfer learning is most effective if the labeled dataset is closely related to the target domain. For instance, if we train a model to classify bird species, we can pretrain a network on a large, general animal classification dataset. However, if such a large animal classification dataset is unavailable, we can also pretrain the model on the relatively broad ImageNet dataset.

A dataset may be extremely small and unsuitable for supervised learning--for example, if it contains only a handful of labeled examples per class. If our classifier needs to operate in a context where the collection of additional labeled data is not feasible, we may also consider **few-shot learning** .

Other Methods

The previous sections covered the main approaches to using and modifying datasets to reduce overfitting. However, this is not an exhaustive list. Other common techniques include:

- Feature engineering and normalization
- The inclusion of adversarial examples and label or feature noise
- Label smoothing
- Smaller batch sizes
- Data augmentation techniques such as Mixup, Cutout, and CutMix

Tips: 减弱 **过度拟合** , 还可以使用下述技术:

- **特征工程** 和 **归一化** : 改进特征选择和标准化数据
- **对抗样本** 和 **标签或特征噪声** : 添加对抗样本或噪声来增强模型的鲁棒性
- **标签平滑** : 软化标签, 避免模型对训练标签过于自信
- **更小的批量大小** : 使用较小的batch size来增加训练的随机性
- **数据增强** 技术, 如 **Mixup** 、 **Cutout** 和 **CutMix**
 - **Mixup** : 混合不同样本的数据
 - **Cutout** : 随机遮挡图像的部分区域

- **CutMix** : 将一张图像的一部分替换为另一张图像的对应部分



The next chapter covers additional techniques to reduce overfitting from a model perspective, and it concludes by discussing which regularization techniques we should consider in practice.

Exercises

5-1. Suppose we train an XGBoost model to classify images based on manually extracted features obtained from collaborators. The dataset of labeled training examples is relatively small, but fortunately, our collaborators also have a labeled training set from an older project on a related domain. We're considering implementing a transfer learning approach to train the XGBoost model. Is this a feasible option? If so, how could we do it? (Assume we are allowed to use only XGBoost and not another classification algorithm or model.)

5-2. Suppose we're working on the image classification problem of implementing MNIST-based handwritten digit recognition. We've added a decent amount of data augmentation to try to reduce overfitting. Unfortunately, we find that the classification accuracy is much worse than it was before the augmentation. What are some potential reasons for this?

References

- Apaperondataaugmentationfortabulardata:DerekSnow, "DeltaPy: A Framework for Tabular Data Augmentation in Python"? (2020), <https://github.com/firmai/deltapy>.
- The paper proposing the GReaT method for generating synthetic tabular data using an auto-regressive generative large language model: Vadim Borisov et al., "Language Models Are Realistic Tabular Data Generators"? (2022), <https://arxiv.org/abs/2210.06280>.
- ThepaperproposingtheTabDDPMmethodforgeneratingsynthetictabulardatausingadiffusionmodel: "Modelling Tabular Data with Diffusion Models"? (2022), <https://arxiv.org/abs/2209.15421>.



- Scikit-learn's user guide offers a section on preprocessing data, featuring techniques like feature scaling and normalization that can enhance your model's performance: <https://scikit-learn.org/stable/modules/preprocessing.html>.
- A survey on methods for robustly training deep models with noisy labels that explores techniques to mitigate the impact of incorrect or misleading target values: Bo Han et al., "A Survey of Label-noise Representation Learning: Past, Present and Future"? (2020), <https://arxiv.org/abs/2011.04406>.
- Theoretical and empirical evidence to support the idea that controlling the ratio of batch size to learning rate in stochastic gradient descent is crucial for good modeling performance in deep neural networks: Fengxiang He, Tongliang Liu, and Dacheng Tao, "Control Batch Size and Learning Rate to Generalize Well: Theoretical and Empirical Evidence"? (2019), <https://dl.acm.org/doi/abs/10.5555/3454287.3454390>.
- Inclusion of adversarial examples, which are input samples designed to mislead the model, can improve prediction performance by making the model more robust: Cihang Xie et al., "Adversarial Examples Improve Image Recognition"? (2019), <https://arxiv.org/abs/1911.09665>.
- Label smoothing is a regularization technique that mitigates the impact of potentially incorrect labels in the dataset by replacing hard 0 and 1 classification targets with softened values: Rafael MÅ¼ller, Simon Kornblith, and Geoffrey Hinton, "When Does Label Smoothing Help?"? (2019), <https://arxiv.org/abs/1906.02629>.
- Mixup, a popular method that trains neural networks on blended data pairs to improve generalization and robustness: Hongyi Zhang et al., "Mixup: Beyond Empirical Risk Minimization"? (2018), <https://arxiv.org/abs/1710.09412>.



Chapter 6: Reducing Overfitting with Model Modifications

Suppose we train a neural network classifier in a supervised fashion and already employ various dataset-related techniques to mitigate overfitting. How can we change the model or make modifications to the training loop to further reduce the effect of overfitting?

The most successful approaches against overfitting include **regularization** techniques like **dropout** and **weight decay**. As a rule of thumb, models with a larger number of parameters require more training data to generalize well. Hence, decreasing the model size and capacity can sometimes also help reduce overfitting. Lastly, building ensemble models is among the most effective ways to combat overfitting, but it comes with increased computational expense.

Tips: 减少 **过拟合**，最有效的技术是 **正则化**，包括 **dropout** 和 **权重衰减**；此外，还可以 **减小模型大小** 和 **构建集成模型**。

This chapter outlines the key ideas and techniques for several categories of reducing overfitting with model modifications and then compares them to one another. It concludes by discussing how to choose between all types of overfitting reduction methods, including those discussed in the previous chapter.

Common Methods

The various model- and training-related techniques to reduce overfitting can be grouped into three broad categories: (1) adding **regularization**, (2) choosing **smaller models**, and (3) building **ensemble models**.

Regularization

We can interpret regularization as a penalty against complexity. Classic regularization techniques for neural networks include L_2 regularization and the related weight decay method. We implement L_2 regularization by adding a penalty term to the loss function that is minimized during training. This added



term represents the size of the weights, such as the squared sum of the weights. The following formula shows an L_2 regularized loss

$$\text{RegularizedLoss} = \text{Loss} + \frac{\lambda}{n} \sum_j w_j^2$$

where λ is a **hyperparameter** that controls the **regularization strength**.

During backpropagation, the optimizer minimizes the modified loss, now including the additional penalty term, which leads to smaller model weights and can improve generalization to unseen data.

Tips: 正则化 **regularization**，通过 **添加惩罚项**，来减少模型的 **权重**。

Weight decay is similar to L_2 regularization but is applied to the optimizer directly rather than modifying the loss function. Since weight decay has the same effect as L_2 regularization, the two methods are often used synonymously, but there may be subtle differences depending on the implementation details and optimizer.

Many other techniques have regularizing effects. For brevity's sake, we'll discuss just two more widely used methods: **dropout** and **early stopping**.

Dropout reduces overfitting by randomly setting some of the activations of the hidden units to zero during training. Consequently, the neural network cannot rely on particular neurons to be activated. Instead, it learns to use a larger number of neurons and multiple independent representations of the same data, which helps to reduce overfitting.

In **early stopping**, we monitor the model's performance on a validation set during training and stop the training process when the performance on the validation set begins to decline, as illustrated in Figure [6.1](#).

Tips: 早停 **early stopping**，通过 **监控模型在验证集上的性能**，来停止训练过程。

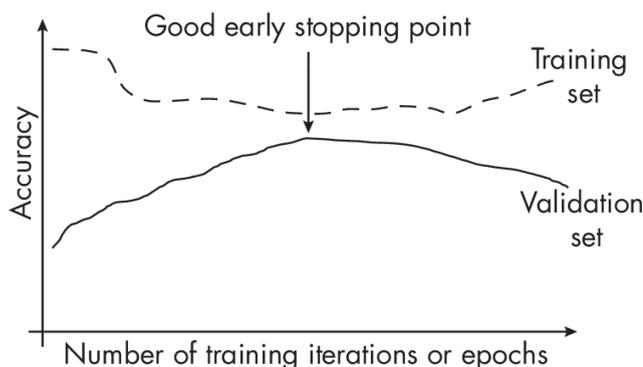


Figure 6.1

In Figure 6.1, we can see that the validation accuracy increases as the training and validation accuracy gap closes. The point where the training and validation accuracy is closest is the point with the least amount of overfitting, which is usually a good point for early stopping.

Smaller Models

Classic bias-variance theory suggests that reducing model size can reduce overfitting. The intuition behind this theory is that, as a general rule of thumb, the smaller the number of model parameters, the smaller its capacity to memorize or overfit to noise in the data. The following paragraphs discuss methods to reduce the model size, including **pruning**, which removes parameters from a model, and **knowledge distillation**, which transfers knowledge to a smaller model.

Tips: 减小模型大小，包括 **剪枝** 和 **知识蒸馏**。

Besides reducing the number of layers and shrinking the layers' widths as a hyperparameter tuning procedure, another approach to obtaining smaller models is **iterative pruning**, in which we train a large model to achieve good performance on the original dataset. We then iteratively remove parameters of the model, retraining it on the dataset such that it maintains the same predictive performance as the original model. (The lottery ticket hypothesis, discussed in Chapter [ch04], uses iterative pruning.)

Tips: 前置减少模型的参数，包括层数和宽度。后置的 **迭代剪枝**，也是常用方法。



Another common approach to obtaining smaller models is **knowledge distillation**. The general idea behind this approach is to transfer knowledge from a large, more complex model (the *teacher*) to a smaller model (the *student*). Ideally, the student achieves the same predictive accuracy as the teacher, but it does so more efficiently due to the smaller size. As a nice side effect, the smaller student may overfit less than the larger teacher model.

Figure 6.2 diagrams the original knowledge distillation process. Here, the **teacher** is first trained in a regular supervised fashion to classify the examples in the dataset well, using a conventional cross-entropy loss between the predicted scores and ground truth class labels. While the smaller **student** network is trained on the same dataset, the training objective is to minimize both

(a) the cross entropy between the outputs and the class labels and

(b) the difference between its outputs and the teacher outputs (measured using *Kullback-Leibler* divergence, which quantifies the difference between two probability distributions by calculating how much one distribution diverges from the other in terms of information content).

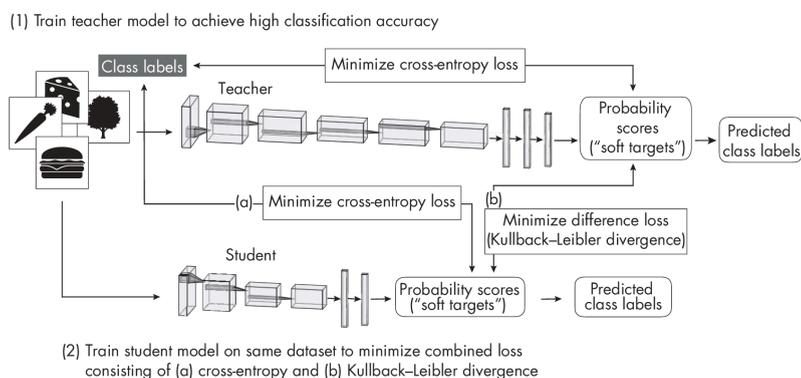


Figure 6.2

By minimizing the Kullback-Leibler divergence--the difference between the teacher and student score distributions--the student learns to mimic the teacher while being smaller and more efficient.

Tips: 知识蒸馏 **knowledge distillation**，通过 **将知识从大模型**，**蒸馏**到 **小模型**，来提高小模型的性能。

Caveats with Smaller Models



While pruning and knowledge distillation can also enhance a model's generalization performance, these techniques are not primary or effective ways of reducing overfitting.

Tips: 剪枝 和 知识蒸馏 ， 可以 提高模型的泛化性能，但 不是 减少过拟合的 主要 方法。

Early research results indicate that pruning and knowledge distillation can improve the generalization performance, presumably due to smaller model sizes. However, counterintuitively, recent research studying phenomena like double descent and grokking also showed that larger, overparameterized models have improved generalization performance if they are trained beyond the point of overfitting. **Double descent** refers to the phenomenon in which models with either a small or an extremely large number of parameters have good generalization performance, while models with a number of parameters equal to the number of training data points have poor generalization performance. *Grokking* reveals that as the size of a dataset decreases, the need for optimization increases, and generalization performance can improve well past the point of overfitting.

Tips: 双降 **double descent** ， 是一种现象，模型参数的数量，在最佳量级之前和之后，都模型泛化效果，都会变差。但是， 涌现/顿悟 **grokking** 现象，展示出当模型参数数量超大时，泛化性能又会变好。 ??? FIXME

How can we reconcile the observation that pruned models can exhibit better generalization performance with contradictory observations from studies of double descent and grokking? Researchers recently showed that the improved training process partly explains the reduction of overfitting due to pruning. Pruning involves more extended training periods and a replay of learning rate schedules that may be partly responsible for the improved generalization performance.

Pruning and knowledge distillation remain excellent ways to improve the computational efficiency of a model. However, while they can also enhance a model's generalization performance, these techniques are not primary or effective ways of reducing overfitting.

Tips: 剪枝和知识蒸馏, 可以提高模型的泛化性能, 但不是减少过拟合的主要方法。



Ensemble Methods

Ensemble methods combine predictions from multiple models to improve the overall prediction performance. However, the downside of using multiple models is an increased computational cost.

We can think of ensemble methods as asking a committee of experts to weigh in on a decision and then combining their judgments in some way to make a final decision. Members in a committee often have different backgrounds and experiences. While they tend to agree on basic decisions, they can overrule bad decisions by majority rule. This doesn't mean that the majority of experts is always right, but there is a good chance that the majority of the committee is more often right, on average, than every single member.

The most basic example of an ensemble method is majority voting. Here, we train k different classifiers and collect the predicted class label from each of these k models for a given input. We then return the most frequent class label as the final prediction. (Ties are usually resolved using a confidence score, randomly picking a label, or picking the class label with the lowest index.)

Ensemble methods are more prevalent in classical machine learning than deep learning because it is more computationally expensive to employ multiple models than to rely on a single one. In other words, deep neural networks require significant computational resources, making them less suitable for ensemble methods.

Random forests and gradient boosting are popular examples of ensemble methods. However, by using majority voting or stacking, for example, we can combine any group of models: an ensemble may consist of a support vector machine, a multilayer perceptron, and a nearest-neighbor classifier. Here, stacking (also known as *stacked generalization*) is a more advanced variant of majority voting that involves training a new model to combine the predictions of several other models rather than obtaining the label by majority vote.

A popular industry technique is to build models from *k-fold cross-validation*, a model evaluation technique in which we train and evaluate a model on k training

folds. We then compute the average performance metric across all k iterations to estimate the overall performance measure of the model. After evaluation, we can either train the model on the entire training dataset or combine the individual models as an ensemble, as shown in Figure 6.2.

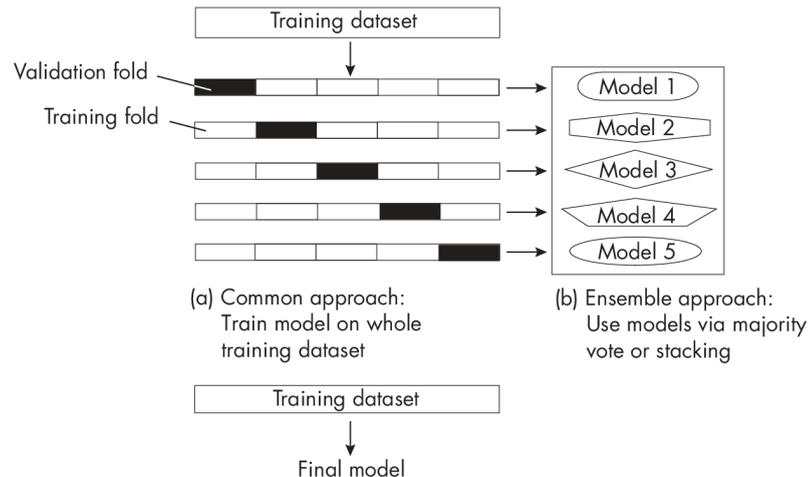


Figure 6.3

As shown in Figure 6.2, the k -fold ensemble approach trains each of the k models on the respective $k - 1$ training folds in each round. After evaluating the models on the validation folds, we can combine them into a majority vote classifier or build an ensemble using stacking, a technique that combines multiple classification or regression models via a meta-model.

While the ensemble approach can potentially reduce overfitting and improve robustness, this approach is not always suitable. For instance, potential downsides include managing and deploying an ensemble of models, which can be more complex and computationally expensive than using a single model.

Other Methods

So far, this book has covered some of the most prominent techniques to reduce overfitting. Chapter [ch05] covered techniques that aim to reduce overfitting from a data perspective. Additional techniques for reducing overfitting with model modifications include skip-connections (found in residual networks, for example), look-ahead optimizers, stochastic weight averaging, multitask learning, and snapshot ensembles.



While they are not originally designed to reduce overfitting, layer input normalization techniques such as batch normalization (BatchNorm) and layer normalization (LayerNorm) can stabilize training and often have a regularizing effect that reduces overfitting. Weight normalization, which normalizes the model weights instead of layer inputs, could also lead to better generalization performance. However, this effect is less direct since weight normalization (WeightNorm) doesn't explicitly act as a regularizer like weight decay does.

Choosing a Regularization Technique

Improving data quality is an essential first step in reducing overfitting. However, for recent deep neural networks with large numbers of parameters, we need to do more to achieve an acceptable level of overfitting. Therefore, data augmentation and pretraining, along with established techniques such as dropout and weight decay, remain crucial overfitting reduction methods.

In practice, we can and should use multiple methods at once to reduce overfitting for an additive effect. To achieve the best results, treat selecting these techniques as a hyperparameter optimization problem.

Exercises

6-1. Suppose we're using early stopping as a mechanism to reduce overfitting-- in particular, a modern early-stopping variant that creates checkpoints of the best model (for instance, the model with the highest validation accuracy) during training so that we can load it after the training has completed. This mechanism can be enabled in most modern deep learning frameworks. However, a colleague recommends tuning the number of training epochs instead. What are some of the advantages and disadvantages of each approach?

6-2. Ensemble models have been established as a reliable and successful method for decreasing overfitting and enhancing the reliability of predictive modeling efforts. However, there's always a trade-off. What are some of the drawbacks associated with ensemble techniques?



References

- For more on the distinction between L_2 regularization and weight decay: Guodong Zhang et al., "Three Mechanisms of Weight Decay Regularization"? (2018), <https://arxiv.org/abs/1810.12281>.
- Research results indicate that pruning and knowledge distillation can improve generalization performance, presumably due to smaller model sizes: Geoffrey Hinton, Oriol Vinyals, and Jeff Dean, "Distilling the Knowledge in a Neural Network"? (2015), <https://arxiv.org/abs/1503.02531>.
- Classic bias-variance theory suggests that reducing model size can reduce overfitting: Jerome H. Friedman, Robert Tibshirani, and Trevor Hastie, "Model Selection and Bias-Variance Tradeoff,"? Chapter 2.9, in *The Elements of Statistical Learning* (Springer, 2009).
- The lottery ticket hypothesis applies knowledge distillation to find smaller networks with the same predictive performance as the original one: Jonathan Frankle and Michael Carbin, "The Lottery Ticket Hypothesis: Finding Sparse, Trainable Neural Networks"? (2018), <https://arxiv.org/abs/1803.03635>.
- For more on double descent: https://en.wikipedia.org/wiki/Double_descent.
- The phenomenon of grokking indicates that generalization performance can improve well past the point of overfitting: Alethea Power et al., "Grokking: Generalization Beyond Overfitting on Small Algorithmic Datasets"? (2022), <https://arxiv.org/abs/2201.02177>.
- Recent research shows that the improved training process partly explains the reduction of overfitting due to pruning: Tian Jin et al., "Pruning's Effect on Generalization Through the Lens of Training and Regularization"? (2022), <https://arxiv.org/abs/2210.13738>.
- Dropout was previously discussed as a regularization technique, but it can also be considered an ensemble method that approximates a weighted geometric mean of multiple networks: Pierre Baldi and Peter J. Sadowski, "Understanding Dropout"? (2013), <https://proceedings.neurips.cc/paper/2013/hash/71f6278d140af599e06ad9bf1ba03cb0-Abstract.html>.

- Regularization cocktails need to be tuned on a per-dataset basis: Arlind Kadra et al., "Well-Tuned Simple Nets Excel on Tabular Datasets"? (2021), <https://arxiv.org/abs/2106.11189>.
-





Chapter 7: Multi-GPU Training Paradigms

What are the different multi-GPU training paradigms, and what are their respective advantages and disadvantages?

Multi-GPU training paradigms can be categorized into two groups: dividing data for parallel processing with multiple GPUs and dividing the model among multiple GPUs to handle memory constraints when the model size surpasses that of a single GPU. Data parallelism falls into the first category, while model parallelism and tensor parallelism fall into the second category. Techniques like pipeline parallelism borrow ideas from both categories. In addition, current software implementations such as DeepSpeed, Colossal AI, and others blend multiple approaches into a hybrid technique.

This chapter introduces several training paradigms and provides advice on which to use in practice.

This chapter primarily uses the term **GPUs** to describe the hardware utilized for parallel processing. However, the same concepts and techniques discussed can be applied to other specialized hardware devices, such as tensor processing units (TPUs) or other accelerators, depending on the specific architecture and requirements of the system.

The Training Paradigms

The following sections discuss the model parallelism, data parallelism, tensor parallelism, and sequence parallelism multi-GPU training paradigms.

Model Parallelism

Model parallelism, or inter-op parallelism, is a technique in which different sections of a large model are placed on different GPUs and are computed sequentially, with intermediate results passed between the devices. This allows for the training and execution of models that might not fit entirely on a single



device, but it can require intricate coordination to manage the dependencies between different parts of the model.

Model parallelism is perhaps the most intuitive form of parallelization across devices. For example, for a simple neural network that consists of only two layers—a hidden layer and an output layer—we can keep one layer on one GPU and the other layer on another GPU. Of course, this can scale to an arbitrary number of layers and GPUs.

This is a good strategy for dealing with limited GPU memory where the complete network does not fit into one GPU. However, there are more efficient ways of using multiple GPUs, such as tensor parallelism, because the chain-like structure (layer 1 on GPU 1 → layer 2 on GPU 2 → ...) in model parallelism introduces a bottleneck. In other words, a major disadvantage of model parallelism is that the GPUs have to wait for each other. They cannot efficiently work in parallel, as they depend on one other's outputs.

Data Parallelism

Data parallelism has been the default mode for multi-GPU training for several years. Here, we divide a minibatch into smaller microbatches. Each GPU then processes a microbatch separately to compute the loss and loss gradients for the model weights. After the individual devices process the microbatches, the gradients are combined to compute the weight update for the next round.

An advantage of data parallelism over model parallelism is that the GPUs can run in parallel. Each GPU processes a portion of the training minibatch, that is, a microbatch. However, a caveat is that each GPU requires a full copy of the model. This is obviously not feasible if we have large models that don't fit into the GPU's VRAM.

Tensor Parallelism

Tensor parallelism, or intra-op parallelism, is a more efficient form of model parallelism. Here, the weight and activation matrices are spread across the devices instead of distributing whole layers across devices: the individual matrices are split, so we split an individual matrix multiplication across GPUs.



We can implement tensor parallelism using basic principles of linear algebra; we can split a matrix multiplication across two GPUs in a row- or column-wise fashion, as illustrated in Figure 7.1 for two GPUs. (This concept can be extended to an arbitrary number of GPUs.)

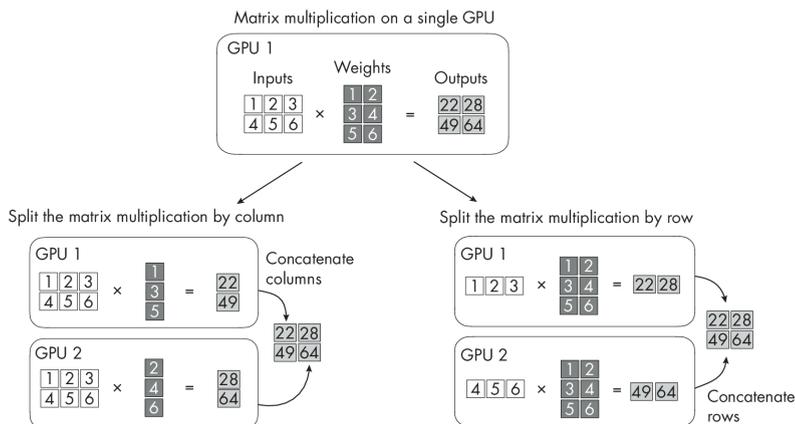


Figure 7.1

Like model parallelism, tensor parallelism allows us to work around memory limitations. At the same time, it also lets us execute operations in parallel, similar to data parallelism.

A small weakness of tensor parallelism is that it can result in high communication overhead between the multiple GPUs across which the matrices are split or sharded. For instance, tensor parallelism requires frequent synchronization of the model parameters across devices, which can slow down the overall training process.

Figure 7.2 compares model, data, and tensor parallelism.

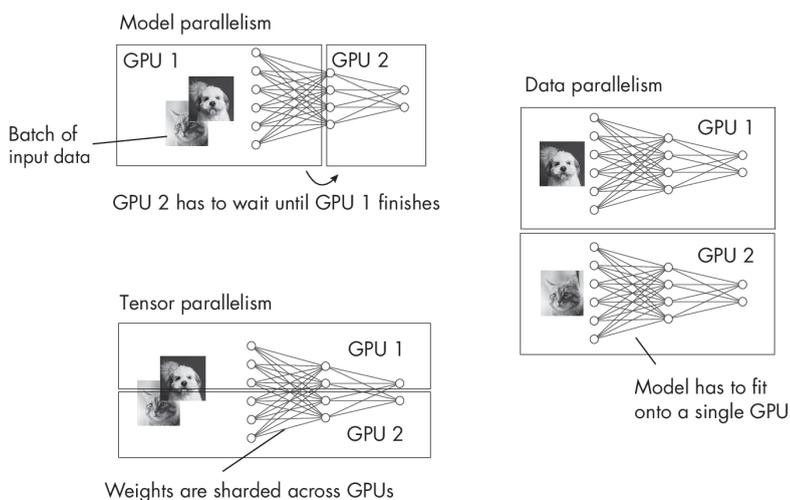


Figure 7.2



In model parallelism, we put different layers onto different GPUs to work around GPU memory limitations. In data parallelism, we split a batch across GPUs to train copies of the model in parallel, averaging gradients for the weight update afterward. In tensor parallelism, we split matrices (inputs and weights) across different GPUs for parallel processing when models are too large to fit into GPU memory.

Pipeline Parallelism

In *pipeline parallelism*, activations are passed during the forward pass, as in model parallelism. The twist is that the gradients of the input tensor are passed backward to prevent the devices from being idle. In a sense, pipeline parallelism is a sophisticated hybrid version of data and model parallelism.

We can think of pipeline parallelism as a form of model parallelism that tries to minimize the sequential computation bottleneck, enhancing the parallelism between the individual layers sitting on different devices. However, pipeline parallelism also borrows ideas from data parallelism, such as splitting minibatches further into microbatches.

Pipeline parallelism is definitely an improvement over model parallelism, though it is not perfect and there will be idle bubbles. A further disadvantage of pipeline parallelism is that it may require significant effort to design and implement the pipeline stages and associated communication patterns. Additionally, the performance gains it generates may not be as substantial as those from other parallelization techniques, such as pure data parallelism, especially for small models or in cases where the communication overhead is high.

For modern architectures that are too large to fit into GPU memory, it is more common nowadays to use a blend of data parallelism and tensor parallelism techniques instead of pipeline parallelism.

Sequence Parallelism

Sequence parallelism aims to address computational bottlenecks when working with long sequences using transformer-based LLMs. More specifically, one shortcoming of transformers is that the self-attention mechanism (the original scaled-dot product attention) scales quadratically with the input sequence length.



There are, of course, more efficient alternatives to the original attention mechanism that scale linearly.

However, these efficient self-attention mechanisms are less popular, and most people still prefer the original scaled-dot product attention mechanism as of this writing. Sequence parallelism, illustrated in Figure 7.3, splits the input sequence into smaller chunks to be distributed across GPUs, which aims to reduce computation memory constraints of self-attention mechanisms.

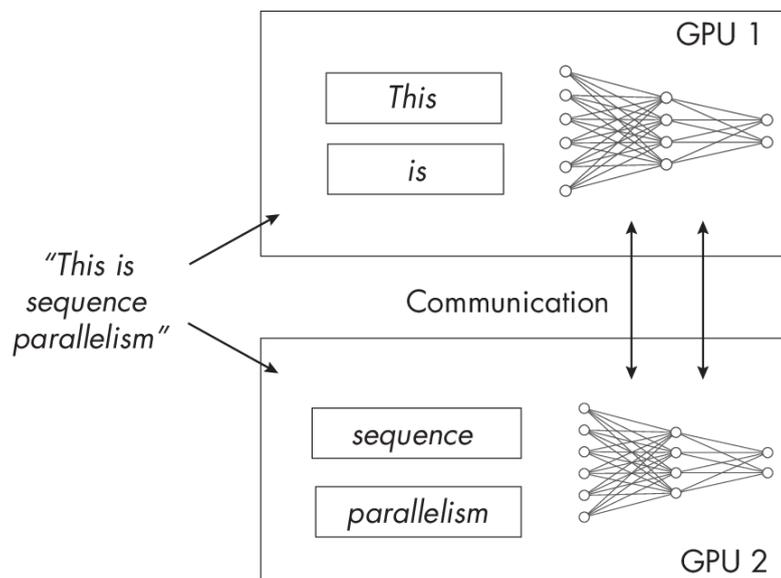


Figure 7.3

How does sequence parallelism relate to the multi-GPU techniques discussed earlier? Sequence parallelism deals specifically with sequential data, tensor parallelism deals with the model's internal structure, and data parallelism deals with how the training data is divided. Theoretically, since each of these parallelism strategies addresses a different aspect of the computational challenge, they can thus be combined in various ways to optimize the training or inference process. Sequence parallelism is not as well studied as other parallelization techniques, however.

While sequence parallelism appears useful in practice, it also introduces additional communication overheads similar to the aforementioned parallelism techniques. Like data parallelism, it requires us to duplicate the model and make sure it fits into the device memory. Another of its disadvantages (depending on the implementation) for multi-GPU training of transformers is that breaking up the input sequence into smaller subsequences can decrease the model's accuracy (mainly when the model is applied to longer sequences).



Recommendations

Practical recommendations depend on the context. If we train small models that fit onto a single GPU, then data parallelism strategies may be the most efficient. Performance gains from pipeline parallelism may not be as significant as those from other parallelization techniques, such as data parallelism, especially for small models or in cases where the communication overhead is high.

If models are too large to fit into the memory of a single GPU, we need to explore model or tensor parallelism. Tensor parallelism is naturally more efficient; the GPUs can work in parallel since there is no sequential dependency as in model parallelism.

Modern multi-GPU strategies also typically combine data parallelism and tensor parallelism.

Exercises

7-1. Suppose we are implementing our own version of tensor parallelism, which works great when we train our model with a standard stochastic gradient descent optimizer. However, when we try the Adam optimizer by Diederik P. Kingma and Jimmy Ba, we encounter an out-of-memory device. What problem might explain this issue?

7-2. Suppose we don't have access to a GPU and are considering using data parallelism on the CPU. Is this a good idea?

References

- The original paper on the Adam optimizer: Diederik P. Kingma and Jimmy Ba, "Adam: A Method for Stochastic Optimization"? (2014), <https://arxiv.org/abs/1412.6980>.
- FormoreonDeepSpeedandColossal-AIformulti-GPUtraining: <https://github.com/microsoft/DeepSpeed> and <https://github.com/hpcaitech/ColossalAI>.



- Pipeline parallelism tutorials and research by the DeepSpeed team: <https://www.deepspeed.ai/tutorials/pipeline> and Yanping Huang et al., "GPipe: Efficient Training of Giant Neural Networks Using Pipeline Parallelism"? (2018), <https://arxiv.org/abs/1811.06965>.
 - The paper proposing sequence parallelism for transformer-based language models: Shenggui Li et al., "Sequence Parallelism: Long Sequence Training from [a] System[s] Perspective"? (2022), <https://arxiv.org/abs/2105.13120>.
 - The scaled-dot product attention mechanism was proposed with the original transformer architecture: Ashish Vaswani et al., "Attention Is All You Need"? (2017), <https://arxiv.org/abs/1706.03762>.
 - A survey covering alternatives to the original self-attention mechanism that scale linearly: Yi Tay et al., "Efficient Transformers: A Survey"? (2020), <https://arxiv.org/abs/2009.06732>.
 - A survey covering additional techniques to improve the training efficiency of transformers: Bohan Zhuang et al., "A Survey on Efficient Training of Transformers"? (2023), <https://arxiv.org/abs/2302.01107>.
 - Modern multi-GPU strategies typically combine data parallelism and tensor parallelism. Popular examples include DeepSpeed stages 2 and 3, described in this tutorial on the zero redundancy optimizer: <https://www.deepspeed.ai/tutorials/zero/>.
-



Chapter 8: The Success of Transformers

What are the main factors that have contributed to the success of transformers?

In recent years, transformers have emerged as the most successful neural network architecture, particularly for various natural language processing tasks. In fact, transformers are now on the cusp of becoming state of the art for computer vision tasks as well. The success of transformers can be attributed to several key factors, including their **attention mechanisms**, ability to be **parallelized easily**, **unsupervised pretraining**, and **high parameter counts**.

Tips:

- 注意力机制，使得模型可以关注到输入序列中的重要部分，从而提高模型性能。
- 模型并行化，从而提高训练速度。
- 自监督预训练，使得模型可以利用大量无标签数据，从而提高模型性能。
- 高参数数量，使得模型可以学习到更复杂的特征，从而提高模型性能。

The Attention Mechanism

The self-attention mechanism found in transformers is one of the key design components that make transformer-based LLMs so successful. However, transformers are not the first architecture to utilize attention mechanisms.

Attention mechanisms were first developed in the context of image recognition back in 2010, before being adopted to aid the translation of long sentences in recurrent neural networks. (Chapter [\[ch16\]](#) compares the attention mechanisms found in recurrent neural networks and transformers in greater detail.)

The aforementioned attention mechanism is inspired by human vision, focusing on specific parts of an image (foveal glimpses) at a time to process information hierarchically and sequentially. In contrast, the fundamental mechanism underlying transformers is a self-attention mechanism used for sequence-to-sequence tasks, such as machine translation and text generation. It allows each



token in a sequence to attend to all other tokens, thus providing context-aware representations of each token.

Tips: 人类视觉系统，是分层处理的，先关注到特定部分（**foveal glimpses** 中央凹注视），然后逐步关注到更多部分(以**分层**和**顺序**的方式处理信息)。

- 人类视觉：像看东西一样，先看重点部分，再逐步看更多细节
- Transformer自注意力：每个词都能"看到"句子中的所有其他词，理解上下文关系

What makes attention mechanisms so unique and useful? For the following illustration, suppose we are using an encoder network on a fixed-length representation of the input sequence or image -- this can be a fully connected, convolutional, or attention-based encoder.

Tips: FIXME, 注意力权重是动态的??? 因为跟元素相对位置有关? 元素就是 token?

- 在Transformer中，编码器使用**自注意力机制**，计算每个输入token相对于序列中其他token的重要性，从而让模型关注输入序列中的相关部分。
- 概念上，注意力机制允许，Transformer关注序列或图像的不同部分。
- 表面上，这听起来非常类似于**全连接层**，其中每个输入元素与下一个层中的输入元素的权重连接。
- 在注意力机制中，计算注意力权重涉及将每个输入元素与所有其他元素进行比较。
- 通过这种方法获得的注意力权重是**动态**的，并且依赖于输入。
- 相比之下，卷积或全连接层的权重在训练后是**固定的**，如Figure 8.1所示。

In a transformer, the encoder uses self-attention mechanisms to compute the importance of each input token relative to other tokens in the sequence, allowing the model to focus on relevant parts of the input sequence. Conceptually, attention mechanisms allow the transformers to attend to different parts of a sequence or image. On the surface, this sounds very similar to a fully connected layer where each input element is connected via a weight with the input element in the next layer. In attention mechanisms, the computation of the attention

weights involves comparing each input element to all others. The attention weights obtained by this approach are dynamic and input dependent. In contrast, the weights of a convolutional or fully connected layer are fixed after training, as illustrated in Figure 8.1.

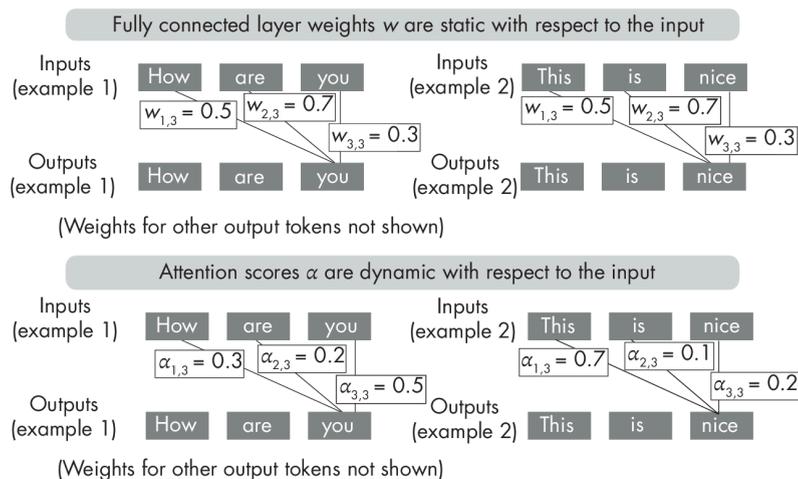


Figure 8.1

As the top part of Figure 8.1 shows, once trained, the weights of fully connected layers remain fixed regardless of the input. In contrast, as shown at the bottom, self-attention weights change depending on the inputs, even after a transformer is trained.

Tips:

- 注意力机制，允许神经网络**选择性**地对**不同输入特征**的重要性进行**加权**，从而让模型专注于给定任务的输入的**最相关部分**。
- 这提供了对每个词或图像token的**上下文理解**，允许更细致的解释，这是使Transformer如此成功的一个方面。

Attention mechanisms allow a neural network to selectively weigh the importance of different input features, so the model can focus on the most relevant parts of the input for a given task. This provides a contextual understanding of each word or image token, allowing for more nuanced interpretations, which is one of the aspects that can make transformers work so well.

Pretraining via Self-Supervised Learning



Tips:

- 自监督预训练，是Transformer成功的一个重要因素。
- 在自监督预训练中，Transformer模型被训练来预测句子中的缺失词或文档中的下一个句子。
- 通过学习预测这些缺失词或下一个句子，模型被迫学习语言的通用表示，可以针对各种下游任务进行微调。

Pretraining transformers via self-supervised learning on large, unlabeled datasets is another key factor in the success of transformers. During pretraining, the transformer model is trained to predict missing words in a sentence or the next sentence in a document, for example. By learning to predict these missing words or the next sentence, the model is forced to learn general representations of language that can be fine-tuned for a wide range of downstream tasks.

While unsupervised pretraining has been highly effective for natural language processing tasks, its effectiveness for computer vision tasks is still an active area of research. (Refer to Chapter [\[ch02\]](#) for a more detailed discussion of self-supervised learning.)

Large Numbers of Parameters

One noteworthy characteristic of transformers is their large model sizes. For example, the popular 2020 GPT-3 model consists of 175 billion trainable parameters, while other transformers, such as switch transformers, have trillions of parameters.

The scale and number of trainable parameters of transformers are essential factors in their modeling performance, particularly for large-scale natural language processing tasks. For instance, [linear scaling laws](#) suggest that the training loss decreases proportionally with an increase in model size, so a doubling of the model size can halve the training loss.

This, in turn, can lead to better performance on the downstream target task. However, it is essential to scale the model size and the number of training tokens equally. This means the number of training tokens should be doubled for every doubling of model size.



Since labeled data is limited, utilizing large amounts of data during unsupervised pretraining is vital.

To summarize, large model sizes and large datasets are critical factors in transformers' success. Additionally, using self-supervised learning, the ability to pretrain transformers is closely tied to using large model sizes and large datasets. This combination has been critical in enabling the success of transformers in a wide range of natural language processing tasks.

Tips: 总而言之，Transformer的成功，很大程度上归功于其 **大模型** 和 **大数据** 的使用。

- 线性缩放定律：训练损失与模型大小成正比，因此增加模型大小可以减少训练损失。
- 训练tokens数量：训练tokens数量应该与模型大小成正比，因此增加模型大小应该增加训练tokens数量。

Easy Parallelization

Training **large models** on **large datasets** requires **vast computational resources** , and it's key that the computations can be parallelized to utilize these resources.

Fortunately, transformers are easy to parallelize since they take a fixed-length sequence of word or image tokens as input. For instance, the self-attention mechanism used in most transformer architectures involves computing the weighted sum between a pair of input elements. Furthermore, these pair-wise token comparisons can be computed independently, as illustrated in Figure [8.2](#), making the self-attention mechanism relatively easy to parallelize across different GPU cores.

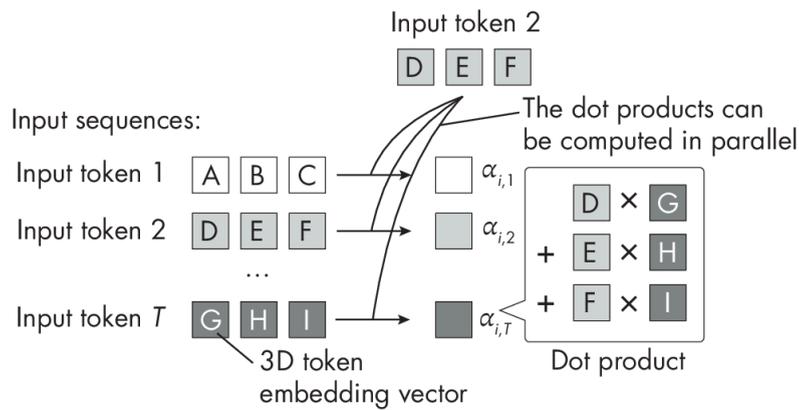


Figure 8.2

In addition, the individual weight matrices used in the self-attention mechanism (not shown in Figure 8.2) can be distributed across different machines for distributed and parallel computing.

Exercises

8-1. As discussed in this chapter, self-attention is easily parallelizable, yet transformers are considered computationally expensive due to self-attention. How can we explain this contradiction?

8-2. Since self-attention scores represent importance weights for the various input elements, can we consider self-attention to be a form of feature selection?

References

- An example of an attention mechanism in the context of image recognition: Hugo Larochelle and Geoffrey Hinton, "Learning to Combine Foveal Glimpses with a Third-Order Boltzmann Machine"? (2010), <https://dl.acm.org/doi/10.5555/2997189.2997328>.
- The paper introducing the self-attention mechanism with the original transformer architecture: Ashish Vaswani et al., "Attention Is All You Need"? (2017), <https://arxiv.org/abs/1706.03762>.
- Transformers can have trillions of parameters: William Fedus, Barret Zoph, and Noam Shazeer, "Switch Transformers: Scaling to Trillion Parameter Models with Simple and Efficient Sparsity"? (2021), <https://arxiv.org/abs/2101.03961>.



- Linear scaling laws suggest that training loss decreases proportionally with an increase in model size: Jared Kaplan et al., "Scaling Laws for Neural Language Models"? (2020), <https://arxiv.org/abs/2001.08361>.
 - Research suggests that in transformer-based language models, the training tokens should be doubled for every doubling of model size: Jordan Hoffmann et al., "Training Compute-Optimal Large Language Models"? (2022), <https://arxiv.org/abs/2203.15556>.
 - Formoreabouttheweightusedinself-attentionandcross-attention mechanisms, check out my blog post: "Understanding and Coding the Self-Attention Mechanism of Large Language Models from Scratch"? at <https://sebastianraschka.com/blog/2023/self-attention-from-scratch.html>.
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Chapter 9: Generative AI Models

What are the popular categories of deep generative models in deep learning (also called *generative AI*), and what are their respective downsides?

Many different types of deep generative models have been applied to generating different types of media: images, videos, text, and audio. Beyond these types of media, models can also be repurposed to generate domain-specific data, such as organic molecules and protein structures. This chapter will first define generative modeling and then outline each type of generative model and discuss its strengths and weaknesses.

Generative vs. Discriminative Modeling

In traditional machine learning, there are two primary approaches to modeling the relationship between input data (x) and output labels (y): **generative models** and **discriminative models**.

- **Generative models** aim to capture the underlying probability distribution of the input data $p(x)$ or the joint distribution $p(x, y)$ between inputs and labels.
- In contrast, **discriminative models** focus on modeling the conditional distribution $p(y | x)$ of the labels given the inputs.

A classic example that highlights the differences between these approaches is to compare the **naive Bayes classifier** and the **logistic regression classifier**.

- Both classifiers estimate the class label probabilities $p(y | x)$ and can be used for classification tasks.
- However, **logistic regression** is considered a discriminative model because it directly models the conditional probability distribution $p(y | x)$ of the class labels given the input features without making assumptions about the underlying joint distribution of inputs and labels.
- **Naive Bayes**, on the other hand, is considered a generative model because it models the joint probability distribution $p(x, y)$ of the input features x and the output labels y . By learning the joint distribution, a generative model like

naive Bayes captures the underlying data generation process, which enables it to generate new samples from the distribution if needed.



Tips:

- 贝叶斯分类器，假设输入和输出之间存在**联合概率分布**；可以生成新的样本，因为它是生成模型；
- 逻辑回归分类器，假设输入和输出之间存在**条件概率分布**；不能生成新的样本；

Types of Deep Generative Models

When we speak of *deep* generative models or deep generative AI, we often loosen this definition to include all types of models capable of producing realistic-looking data (typically text, images, videos, and sound). The remainder of this chapter briefly discusses the different types of deep generative models used to generate such data.

Energy-Based Models

Energy-based models (EBMs) are a class of generative models that learn an energy function, which assigns a scalar value (energy) to each data point. Lower energy values correspond to more likely data points. The model is trained to minimize the energy of real data points while increasing the energy of generated data points.

Examples of EBMs include **deep Boltzmann machines (DBMs)**.

One of the early breakthroughs in deep learning, DBMs provide a means to learn complex representations of data. You can think of them as a form of unsupervised pretraining, resulting in models that can then be fine-tuned for various tasks.

Somewhat similar to naive Bayes and logistic regression, DBMs and multilayer perceptrons (MLPs) can be thought of as generative and discriminative counterparts, with DBMs focusing on capturing the data generation process and MLPs focusing on modeling the decision boundary between classes or mapping inputs to outputs.

A **DBM** consists of multiple layers of hidden nodes, as shown in Figure 9.1. As the figure illustrates, along with the hidden layers, there's usually a visible layer that corresponds to the observable data. This visible layer serves as the input layer where the actual data or features are fed into the network. In addition to using a different learning algorithm than MLPs (contrastive divergence instead of backpropagation), DBMs consist of binary nodes (neurons) instead of continuous ones.

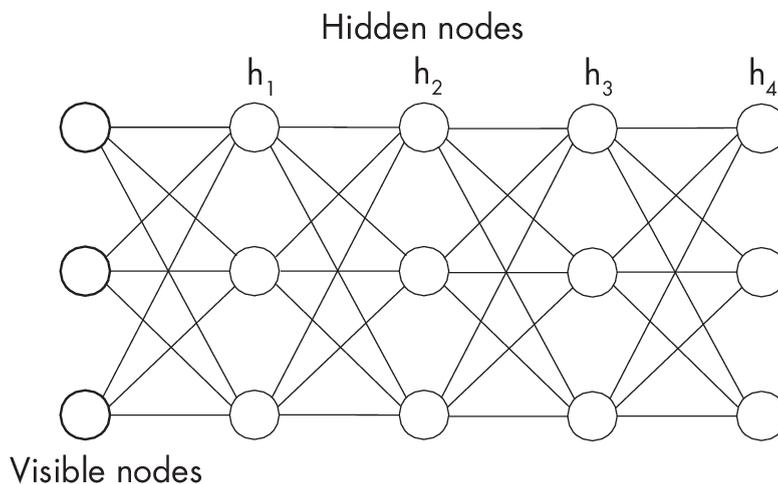


Figure 9.1

Suppose we are interested in generating images. A DBM can learn the joint probability distribution over the pixel values in a simple image dataset like MNIST. To generate new images, the DBM then samples from this distribution by performing a process called *Gibbs sampling*. Here, the visible layer of the DBM represents the input image. To generate a new image, the DBM starts by initializing the visible layer with random values or, alternatively, uses an existing image as a seed. Then, after completing several Gibbs sampling iterations, the final state of the visible layer represents the generated image.

DBMs played an important historical role as one of the first deep generative models, but they are no longer very popular for generating data. They are expensive and more complicated to train, and they have lower expressivity compared to the newer models described in the following sections, which generally results in lower-quality generated samples.

Variational Autoencoders



Variational autoencoders (VAEs) are built upon the principles of variational inference and autoencoder architectures. *Variational inference* is a method for approximating complex probability distributions by optimizing a simpler, tractable distribution to be as close as possible to the true distribution. *Autoencoders* are unsupervised neural networks that learn to compress input data into a low-dimensional representation (encoding) and subsequently reconstruct the original data from the compressed representation (decoding) by minimizing the reconstruction error.

The VAE model consists of two main submodules: an encoder network and a decoder network. The encoder network takes, for example, an input image and maps it to a latent space by learning a probability distribution over the latent variables. This distribution is typically modeled as a Gaussian with parameters (mean and variance) that are functions of the input image. The decoder network then takes a sample from the learned latent distribution and reconstructs the input image from this sample. The goal of the VAE is to learn a compact and expressive latent representation that captures the essential structure of the input data while being able to generate new images by sampling from the latent space. (See Chapter [\[ch01\]](#) for more details on latent representations.)

Figure [9.2](#) illustrates the encoder and decoder submodules of an auto-encoder, where x' represents the reconstructed input x . In a standard variational autoencoder, the latent vector is sampled from a distribution that approximates a standard Gaussian distribution.

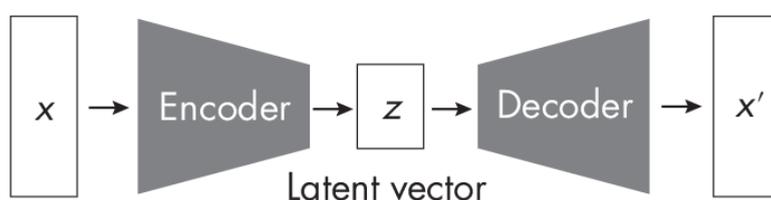


Figure 9.2

Training a VAE involves optimizing the model's parameters to minimize a loss function composed of two terms: a reconstruction loss and a Kullback -- Leibler-divergence (KL-divergence) regularization term. The reconstruction loss ensures that the decoded samples closely resemble the input images, while the KL-divergence term acts as a surrogate loss that encourages the learned latent distribution to be close to a predefined prior distribution (usually a standard Gaussian). To generate new images, we then sample points from the latent



space's prior (standard Gaussian) distribution and pass them through the decoder network, which generates new, diverse images that look similar to the training data.

Disadvantages of VAEs include their complicated loss function consisting of separate terms, as well as their often low expressiveness. The latter can result in blurrier images compared to other models, such as generative adversarial networks.

Generative Adversarial Networks

Generative adversarial networks (GANs) are models consisting of interacting subnetworks designed to generate new data samples that are similar to a given set of input data. While both GANs and VAEs are latent variable models that generate data by sampling from a learned latent space, their architectures and learning mechanisms are fundamentally different.

GANs consist of two neural networks, a generator and a discriminator, that are trained simultaneously in an adversarial manner. The generator takes a random noise vector from the latent space as input and generates a synthetic data sample (such as an image). The discriminator's task is to distinguish between real samples from the training data and fake samples generated by the generator, as illustrated in Figure 9.3.

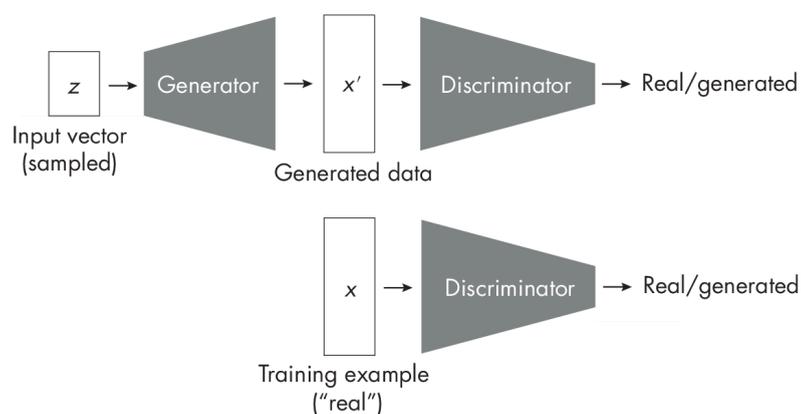


Figure 9.3

The generator in a GAN somewhat resembles the decoder of a VAE in terms of its functionality. During inference, both GAN generators and VAE decoders take random noise vectors sampled from a known distribution (for example, a standard Gaussian) and transform them into synthetic data samples, such as images.



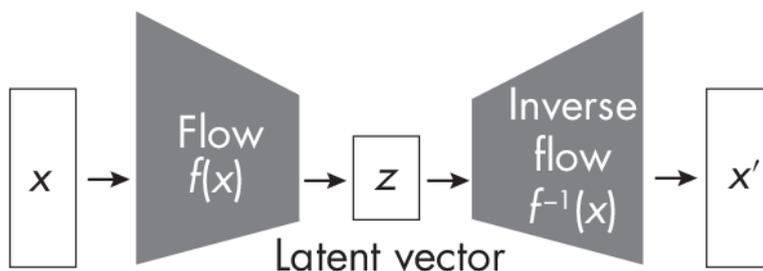
One significant disadvantage of GANs is their unstable training due to the adversarial nature of the loss function and learning process. Balancing the learning rates of the generator and discriminator can be difficult and can often result in oscillations, mode collapse, or non-convergence. The second main disadvantage of GANs is the low diversity of their generated outputs, often due to mode collapse. Here, the generator is able to fool the discriminator successfully with a small set of samples, which are representative of only a small subset of the original training data.

Flow-Based Models

The core concept of *flow-based models*, also known as *normalizing flows*, is inspired by long-standing methods in statistics. The primary goal is to transform a simple probability distribution (like a Gaussian) into a more complex one using invertible transformations.

Although the concept of **normalizing flows** has been apart of the statistics field for a long time, the implementation of early flow-based deep learning models, particularly for image generation, is a relatively recent development. One of the pioneering models in this area was the *non-linear independent components estimation (NICE)* approach. NICE begins with a simple probability distribution, often something straightforward like a normal distribution. You can think of this as a kind of "random noise," or data with no particular shape or structure. NICE then applies a series of transformations to this simple distribution. Each transformation is designed to make the data look more like the final target (for instance, the distribution of real-world images). These transformations are "invertible," meaning we can always reverse them back to the original simple distribution. After several successive transformations, the simple distribution has morphed into a complex distribution that closely matches the distribution of the target data (such as images). We can now generate new data that looks like the target data by picking random points from this complex distribution.

Figure 9.4 illustrates the concept of a flow-based model, which maps the complex input distribution to a simpler distribution and back.



At first glance, the illustration is very similar to the VAE illustration in Figure 9.2. However, while VAEs use neural network encoders like convolutional neural networks, the flow-based model uses simpler decoupling layers, such as simple linear transformations. Additionally, while the decoder in a VAE is independent of the encoder, the data-transforming functions in the flow-based model are mathematically inverted to obtain the outputs.

Unlike VAEs and GANs, flow-based models provide exact likelihoods, which gives us insights into how well the generated samples fit the training data distribution. This can be useful in anomaly detection or density estimation, for example. However, the quality of flow-based models for generating image data is usually lower than GANs. Flow-based models also often require more memory and computational resources than GANs or VAEs since they must store and compute inverses of transformations.

Autoregressive Models

Autoregressive models are designed to predict the next value based on current (and past) values. LLMs for text generation, like ChatGPT (discussed further in Chapter [ch17]), are one popular example of this type of model.

Similar to generating one word at a time, in the context of image generation, autoregressive models like `Pixe1CNN` try to predict one pixel at a time, given the pixels they have seen so far. Such a model might predict pixels from top left to bottom right, in a raster scan order, or in any other defined order.

To illustrate how autoregressive models generate an image one pixel at a time, suppose we have an image of size $H \times W$ (where H is the height and W is the width), ignoring the color channel for simplicity's sake. This image consists of N pixels, where $i = 1, \dots, N$. The probability of observing a particular image in



the dataset is then $P(\text{Image}) = P(i_1, i_2, \dots, i_N)$. Based on the chain rule of probability in statistics, we can decompose this joint probability into conditional probabilities:

$$\begin{aligned} P(\text{Image}) &= P(i_1, i_2, \dots, i_N) \\ &= P(i_1) \cdot P(i_2 | i_1) \cdot P(i_3 | i_1, i_2) \dots P(i_N | i_1 \dots i_{N-1}) \end{aligned}$$

Here, $P(i_1)$ is the probability of the first pixel, $P(i_2|i_1)$ is the probability of the second pixel given the first pixel, $P(i_3|i_1, i_2)$ is the probability of the third pixel given the first and second pixels, and so on.

In the context of image generation, an autoregressive model essentially tries to predict one pixel at a time, as described earlier, given the pixels it has seen so far.

Figure 9.5 illustrates this process, where pixels i_1, \dots, i_{53} represent the context and pixel i_{54} is the next pixel to be generated.

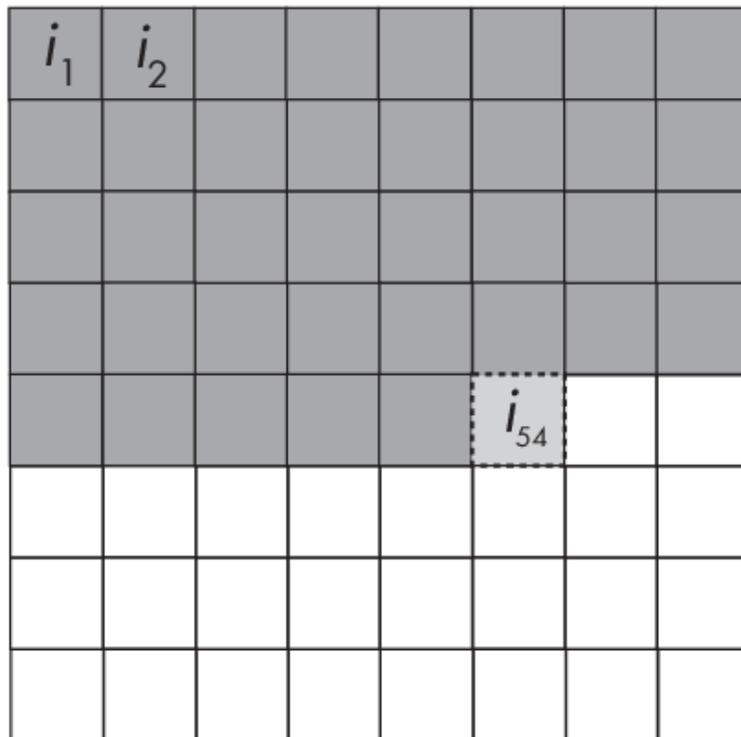


Figure 9.5

The advantage of autoregressive models is that the next-pixel (or word) prediction is relatively straightforward and interpretable. In addition, autoregressive models can compute the likelihood of data exactly, similar to flow-based models, which can be useful for tasks like anomaly detection. Furthermore, autoregressive



models are easier to train than GANs as they don't suffer from issues like mode collapse and other training instabilities.

However, autoregressive models can be slow at generating new samples. This is because they have to generate data one step at a time (for example, pixel by pixel for images), which can be computationally expensive. Autoregressive models may also struggle to capture long-range dependencies because each output is conditioned only on previously generated outputs.

In terms of overall image quality, autoregressive models are therefore usually worse than GANs but are easier to train.

Diffusion Models

As discussed in the previous section, flow-based models transform a simple distribution (such as a standard normal distribution) into a complex one (the target distribution) by applying a sequence of invertible and differentiable transformations (flows). Like flow-based models, *diffusion models* also apply a series of transformations. However, the underlying concept is fundamentally different.

Diffusion models transform the input data distribution into a simple noise distribution over a series of steps using stochastic differential equations. Diffusion is a stochastic process in which noise is progressively added to the data until it resembles a simpler distribution, like Gaussian noise. To generate new samples, the process is then reversed, starting from noise and progressively removing it.

Figure 9.6 outlines the process of adding and removing Gaussian noise from an input image x . During inference, the reverse diffusion process is used to generate a new image x , starting with the noise tensor z_T sampled from a Gaussian distribution.

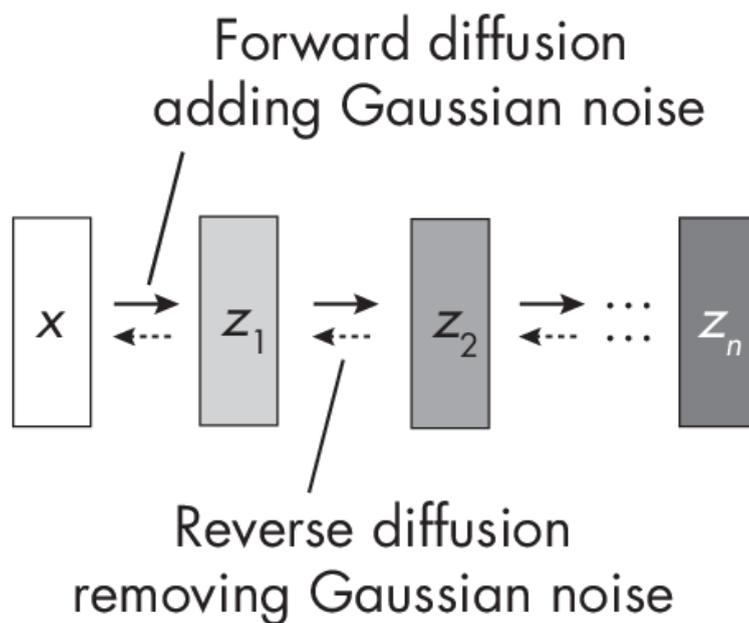


Figure 9.6

While both diffusion models and flow-based models are generative models aiming to learn complex data distributions, they approach the problem from different angles. Flow-based models use deterministic invertible transformations, while diffusion models use the aforementioned stochastic diffusion process.

Recent projects have established state-of-the-art performance in generating high-quality images with realistic details and textures. Diffusion models are also easier to train than GANs. The downside of diffusion models, however, is that they are slower to sample from since they require running a series of sequential steps, similar to flow-based models and autoregressive models. This can make diffusion models less practical for some applications requiring fast sampling.

Consistency Models

Consistency models train a neural network to map a noisy image to a clean one. The network is trained on a dataset of pairs of noisy and clean images and learns to identify patterns in the clean images that are modified by noise. Once the network is trained, it can be used to generate reconstructed images from noisy images in one step.

Consistency model training employs an *ordinary differential equation (ODE)* trajectory, a path that a noisy image follows as it is gradually denoised. The ODE

trajectory is defined by a set of differential equations that describe how the noise in the image changes over time, as illustrated in Figure 9.7.

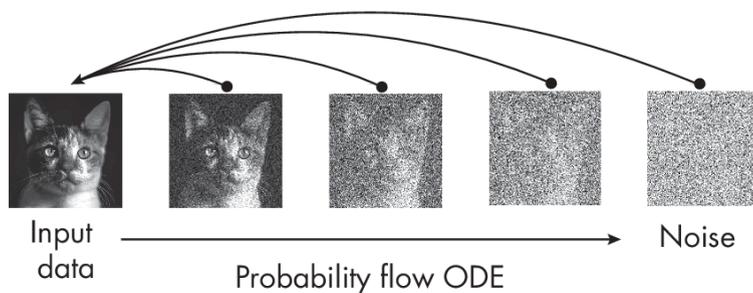


Figure 9.7

As Figure 9.7 demonstrates, we can think of consistency models as models that learn to map any point from a probability flow ODE, which smoothly converts data to noise, to the input.

At the time of writing, consistency models are the most recent type of generative AI model. Based on the original paper proposing this method, consistency models rival diffusion models in terms of image quality. Consistency models are also faster than diffusion models because they do not require an iterative process to generate images; instead, they generate images in a single step.

However, while consistency models allow for faster inference, they are still expensive to train because they require a large dataset of pairs of noisy and clean images.

Recommendations

Deep Boltzmann machines are interesting from a historical perspective since they were one of the pioneering models to effectively demonstrate the concept of unsupervised learning. Flow-based and autoregressive models may be useful when you need to estimate exact likelihoods. However, other models are usually the first choice when it comes to generating high-quality images.

In particular, VAEs and GANs have competed for years to generate the best high-fidelity images. However, in 2022, diffusion models began to take over image generation almost entirely. Consistency models are a promising alternative to diffusion models, but it remains to be seen whether they become more widely adopted to generate state-of-the-art results. The trade-off here is that sampling

from diffusion models is generally slower since it involves a sequence of noise-removal steps that must be run in order, similar to autoregressive models. This can make diffusion models less practical for some applications requiring fast sampling.



Exercises

9-1. How would we evaluate the quality of the images generated by a generative AI model?

9-2. Given this chapter's description of consistency models, how would we use them to generate new images?

References

- The original paper proposing variational autoencoders: Diederik P. Kingma and Max Welling, "Auto-Encoding Variational Bayes"? (2013), <https://arxiv.org/abs/1312.6114>.
- The paper introducing generative adversarial networks: Ian J. Goodfellow et al., "Generative Adversarial Networks"? (2014), <https://arxiv.org/abs/1406.2661>.
- The paper introducing NICE: Laurent Dinh, David Krueger, and Yoshua Bengio, "NICE: Non-linear Independent Components Estimation"? (2014), <https://arxiv.org/abs/1410.8516>.
- The paper proposing the autoregressive PixelCNN model: Aaron van den Oord et al., "Conditional Image Generation with PixelCNN Decoders"? (2016), <https://arxiv.org/abs/1606.05328>.
- The paper introducing the popular Stable Diffusion latent diffusion model: Robin Rombach et al., "High-Resolution Image Synthesis with Latent Diffusion Models"? (2021), <https://arxiv.org/abs/2112.10752>.
- The Stable Diffusion code implementation: <https://github.com/CompVis/stable-diffusion>.

- The paper originally proposing consistency models: Yang Song et al., "Consistency Models"? (2023), <https://arxiv.org/abs/2303.01469>.
-





Chapter 10: Sources of Randomness

What are the common sources of randomness when training deep neural networks that can cause non-reproducible behavior during training and inference?

When training or using machine learning models such as deep neural networks, several sources of randomness can lead to different results every time we train or run these models, even though we use the same overall settings. Some of these effects are accidental and some are intended. The following sections categorize and discuss these various sources of randomness.

Tips:

- 在训练和使用机器学习模型时，如深度神经网络，**随机性**会导致每次 **训练** 或 **运行** 模型时得到不同的结果，即使我们使用相同的配置。
- 这些随机性可能是 **偶然** 的，也可能是 **故意** 的。

Optional hands-on examples for most of these categories are provided in the *supplementary/q10-random-sources* subfolder at <https://github.com/rasbt/MachineLearning-QandAI-book>.

Model Weight Initialization

All common deep neural network frameworks, including TensorFlow and PyTorch, randomly initialize the weights and bias units at each layer by default. This means that the final model will be different every time we start the training. The reason these trained models will differ when we start with different random weights is the nonconvex nature of the loss, as illustrated in Figure [10.1](#). As the figure shows, the loss will converge to different local minima depending on where the initial starting weights are located.

Tips: 初始化权重，会导致不同的局部最优解。

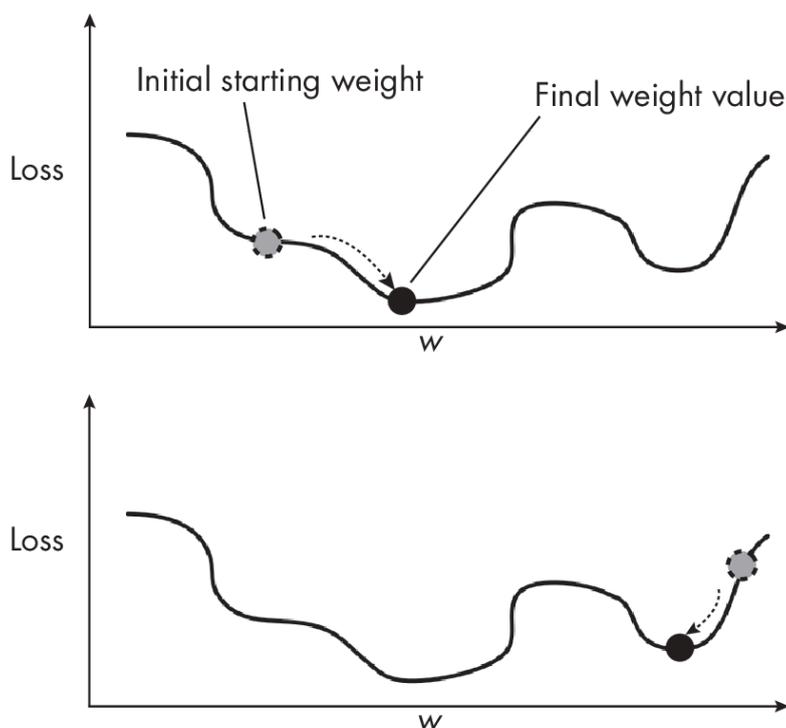


Figure 10.1

In practice, it is therefore recommended to run the training (if the computational resources permit) at least a handful of times; unlucky initial weights can sometimes cause the model not to converge or to converge to a local minimum corresponding to poorer predictive accuracy.

Tips: 实践中，建议 **至少** 运行 **训练几次**，以避免不幸运的初始权重，导致模型不收敛或收敛到较差的局部最优解。

However, we can make the random weight initialization deterministic by seeding the random generator. For instance, if we set the seed to a specific value like 123, the weights will still initialize with small random values. Nonetheless, the neural network will consistently initialize with the **same small random weights**, enabling accurate reproduction of results.

Tips: 通过设置 **随机种子**，可以使得 **初始化** 权重参数是 **确定的**。

Dataset Sampling and Shuffling

When we train and evaluate machine learning models, we usually start by dividing a dataset into training and test sets. This requires random sampling since we have



to decide which examples we put into a training set and which examples we put into a test set.

In practice, we often use model evaluation techniques such as k -fold cross-validation or holdout validation. In holdout validation, we split the training set into **training**, **validation**, and **test** datasets, which are also sampling procedures influenced by randomness. Similarly, unless we use a fixed random seed, we get a different model each time we partition the dataset or tune or evaluate the model using k -fold cross-validation since the training partitions will differ.

Tips:

- 在训练和评估机器学习模型时，我们通常将数据集分为 **训练集**、**验证集** 和 **测试集**。
- 这需要 **随机采样**，因为我们必须决定将哪些样本放入训练集，哪些放入验证集，哪些放入测试集。
- 除非我们使用固定的随机种子，否则每次划分数据集或使用 **k** 折交叉验证时，我们都会得到不同的模型。

Nondeterministic Algorithms

We may include random components and algorithms depending on the architecture and hyperparameter choices. A popular example of this is **dropout**.

Dropout works by randomly setting a fraction of a layer's units to zero during training, which helps the model learn more robust and generalized representations. This "**dropping out**" is typically applied at each training iteration with a probability p , a hyperparameter that controls the fraction of units dropped out. Typical values for p are in the range of 0.2 to 0.8.

To illustrate this concept, Figure [10.2](#) shows a small neural network where dropout randomly drops a subset of the hidden layer nodes in each forward pass during training.

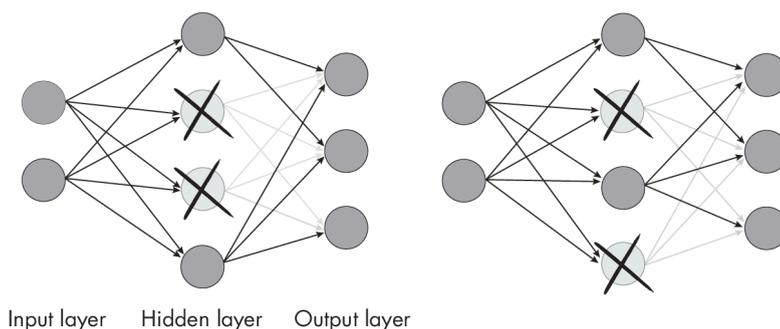


Figure 10.2

To create reproducible training runs, we must seed the random generator before training with dropout (analogous to seeding the random generator before initializing the model weights). During inference, we need to disable dropout to guarantee deterministic results. Each deep learning framework has a specific setting for that purpose -- a PyTorch example is included in the *supplementary/q10-random-sources* subfolder at <https://github.com/rasbt/MachineLearning-QandAI-book>.

Tips:

- 为了创建可复现的训练运行，我们必须在训练前设置随机种子（类似于在初始化模型权重之前设置随机种子）。
- 在推理时，我们需要禁用 `dropout`，以保证结果的确定性。
- 每个深度学习框架都有特定的设置，以实现这一点。

Different Runtime Algorithms

The most intuitive or simplest implementation of an algorithm or method is not always the best one to use in practice. For example, when training deep neural networks, we often use efficient alternatives and approximations to gain speed and resource advantages during training and inference.

A popular example is the convolution operation used in convolutional neural networks. There are several possible ways to implement the convolution operation:

The `classic direct convolution` The common implementation of discrete convolution via an element-wise product between the input and the window,



followed by summing the result to get a single number. (See Chapter [\[ch12\]](#) for a discussion of the convolution operation.)

FFT-based convolution Uses **fast Fourier transform** (FFT) to convert the convolution into an element-wise multiplication in the frequency domain.

Winograd-based convolution An efficient algorithm for small filter sizes (like 3×3 that reduces the number of multiplications required for the convolution.

Different convolution algorithms have different trade-offs in terms of memory usage, computational complexity, and speed. By default, libraries such as the **CUDA Deep Neural Network library** (**cuDNN**), which are used in PyTorch and TensorFlow, can choose different algorithms for performing convolution operations when running deep neural networks on GPUs. However, the deterministic algorithm choice has to be explicitly enabled. In PyTorch, for example, this can be done by setting

```
torch.use_deterministic_algorithms(True)
```

While these approximations yield similar results, subtle numerical differences can accumulate during training and cause the training to converge to slightly different local minima.

Tips: 算法自身也会带来随机性，特别是不同优化算法实现，本身得到的就是近似效果。

- **不同的卷积算法**在 **内存** 使用、 **计算复杂度** 和 **速度** 方面有不同的权衡。
- 默认情况下，PyTorch和TensorFlow等库中的 **CUDA Deep Neural Network library** (**cuDNN**) 可以选择不同的算法来执行卷积操作。
- 但是，确定性算法的选择必须显式启用。
- 在PyTorch中，可以通过设置 `torch.use_deterministic_algorithms(True)` 来启用确定性算法。

Hardware and Drivers



Training deep neural networks on different hardware can also produce different results due to small numeric differences, even when the same algorithms are used and the same operations are executed. These differences may sometimes be due to different numeric precision for floating-point operations. However, small numeric differences may also arise due to hardware and software optimization, even at the same precision.

Tips: 硬件和驱动也会带来随机性，特别是不同硬件平台，不同优化库，不同优化算法实现，本身得到的就是近似效果。

- 不同的 **数值精度**，会导致不同的结果。
- 不同的 **硬件** 和 **软件** 优化，会导致不同的结果。

For instance, different hardware platforms may have specialized optimizations or libraries that can slightly alter the behavior of deep learning algorithms. To give one example of how different GPUs can produce different modeling results, the following is a quotation from the official NVIDIA documentation: "Across different architectures, no cuDNN routines guarantee bit-wise reproducibility. For example, there is no guarantee of bit-wise reproducibility when comparing the same routine run on NVIDIA *Volta*TM and NVIDIA *Turing*TM [. . .] and NVIDIA Ampere architecture."?

Tips:

- 不同的硬件平台可能具有专门的优化或库，可以稍微改变深度学习算法的性能。
- 例如，不同的GPU可以产生不同的建模结果。

Randomness and Generative AI

Besides the various sources of randomness mentioned earlier, certain models may also exhibit random behavior during inference that we can think of as "**randomness by design**."? For instance, generative image and language models may create different results for identical prompts to produce a diverse sample of results. For image models, this is often so that users can select the most accurate and aesthetically pleasing image. For language models, this is often to vary the responses, for example, in chat agents, to avoid repetition.



Tips:

- 除了前面提到的各种随机性来源，某些模型在**推理时**也可能表现出**随机行为**，我们可以将其视为“**设计上的随机性**”。
- 例如，生成式图像和语言模型可能会对相同的提示产生不同的结果，以产生多样化的结果样本。
- 对于图像模型，这通常是为了让用户选择最准确和最吸引人的图像。
- 对于语言模型，这通常是为了避免重复，例如在聊天代理中。

The intended randomness in generative image models during inference is often due to sampling different noise values at each step of the reverse process. In diffusion models, a noise schedule defines the noise variance added at each step of the diffusion process.

Tips:

- 在生成式图像模型中，推理时的随机性，通常是由于在反向过程中对不同的**噪声值**进行采样。
- 在扩散模型中，**噪声调度**定义了扩散过程中添加的**噪声方差**。

Autoregressive LLMs like GPT tend to create different outputs for the same input prompt (GPT will be discussed at greater length in Chapters [\[ch14\]](#) and [\[ch17\]](#)). The ChatGPT user interface even has a Regenerate Response button for that purpose. The ability to generate different results is due to the sampling strategies these models employ. Techniques such as top- k sampling, nucleus sampling, and temperature scaling influence the model's output by controlling the degree of randomness. This is a feature, not a bug, since it allows for diverse responses and prevents the model from producing overly deterministic or repetitive outputs. (See Chapter [\[ch09\]](#) for a more in-depth overview of generative AI and deep learning models; see Chapter [\[ch17\]](#) for more detail on autoregressive LLMs.)

Tips:

- 自回归语言模型（如GPT）倾向于对相同的输入提示，产生不同的输出。
- 这是因为这些模型，采用了不同的**采样策略**。
- 例如，**top- k 采样**、**核采样**和**温度缩放**等技术，通过控制随机性程度，影响模型的输出。

Top- k sampling, illustrated in Figure 10.3, works by sampling tokens from the top k most probable candidates at each step of the next-word generation process.

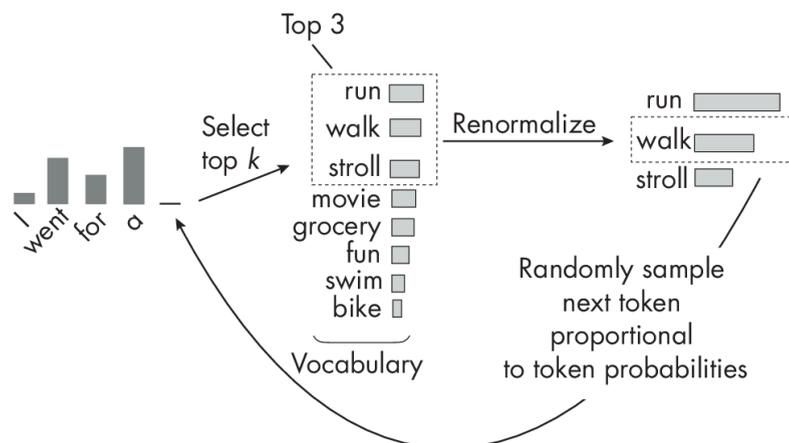


Figure 10.3

Given an input prompt, the language model produces a probability distribution over the entire vocabulary (the candidate words) for the next token. Each token in the vocabulary is assigned a probability based on the model's understanding of the context. The selected top- k tokens are then renormalized so that the probabilities sum to 1. Finally, a token is sampled from the renormalized top- k probability distribution and is appended to the input prompt. This process is repeated for the desired length of the generated text or until a stop condition is met.

Nucleus sampling (also known as *top- p sampling*), illustrated in Figure 10.4, is an alternative to top- k sampling.

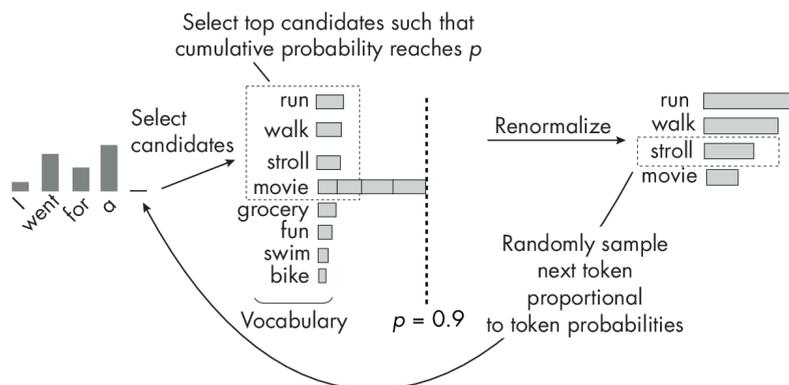


Figure 10.4

Similar to top- k sampling, the goal of nucleus sampling is to balance diversity and coherence in the output. However, nucleus and top- k sampling differ in how to



select the candidate tokens for sampling at each step of the generation process. Top- k sampling selects the k most probable tokens from the probability distribution produced by the language model, regardless of their probabilities. The value of k remains fixed throughout the generation process. Nucleus sampling, on the other hand, selects tokens based on a probability threshold p , as shown in Figure 10.4. It then accumulates the most probable tokens in descending order until their cumulative probability meets or exceeds the threshold p . In contrast to top- k sampling, the size of the candidate set (nucleus) can vary at each step.

Tips:

- 与top- k 采样类似，核采样的目标是平衡输出中的多样性和连贯性。
- 然而，核采样和top- k 采样在选择每个生成步骤中的候选标记时有所不同。
- **top- k 采样***从语言模型产生的概率分布中选择 **概率最高** 的 k *个标记，而核采样**则根据 **累计的概率阈值** p 选择标记。

Exercises

10-1. Suppose we train a neural network with top- k or nucleus sampling where k and p are hyperparameter choices. Can we make the model behave deterministically during inference without changing the code?

10-2. In what scenarios might random dropout behavior during inference be desired?

References

- For more about different data sampling and model evaluation techniques, see my article: "Model Evaluation, Model Selection, and Algorithm Selection in Machine Learning"? (2018), <https://arxiv.org/abs/1811.12808>.
- The paper that originally proposed the dropout technique: Nitish Srivastava et al., "Dropout: A Simple Way to Prevent Neural Networks from Overfitting"? (2014), <https://jmlr.org/papers/v15/srivastava14a.html>.



- A detailed paper on FFT-based convolution: Lu Chi, Borui Jiang, and Yadong Mu, "Fast Fourier Convolution"? (2020),
<https://dl.acm.org/doi/abs/10.5555/3495724.3496100>.
 - Details on Winograd-based convolution: Syed Asad Alam et al., "Winograd Convolution for Deep Neural Networks: Efficient Point Selection"? (2022),
<https://arxiv.org/abs/2201.10369>.
 - More information about the deterministic algorithm settings in PyTorch:
https://pytorch.org/docs/stable/generated/torch.use_deterministic_algorithms.html.
 - For details on the deterministic behavior of NVIDIA graphics cards, see the "Reproducibility"? section of the official NVIDIA documentation:
<https://docs.nvidia.com/deeplearning/cudnn/developer-guide/index.html#reproducibility>.
-



Chapter 11: Calculating the Number of Parameters

How do we compute the number of parameters in a convolutional neural network, and why is this information useful?

Knowing the number of parameters in a model helps gauge the model's size, which affects storage and memory requirements. The following sections will explain how to compute the convolutional and fully connected layer parameter counts.

Tips: 模型参数的数量，是衡量模型大小（存储空间大小）的重要指标，用于估算所需的存储空间。重点关注 **卷积层** 和 **全连接层**。

How to Find Parameter Counts

Suppose we are working with a convolutional network that has two **convolutional layers** with kernel size 5 and kernel size 3, respectively.

- The first convolutional layer has 3 input channels and 5 output channels,
- and the second one has 5 input channels and 12 output channels.
- The stride of these **convolutional layers** is 1.

Furthermore, the network has two **pooling layers**,

- one with a kernel size of 3 and a stride of 2,
- and another with a kernel size of 5 and a stride of 2.

It also has two **fully connected hidden layers** with 192 and 128 hidden units each, where the output layer is a **classification layer** for 10 classes.

The architecture of this network is illustrated in Figure [11.1](#).

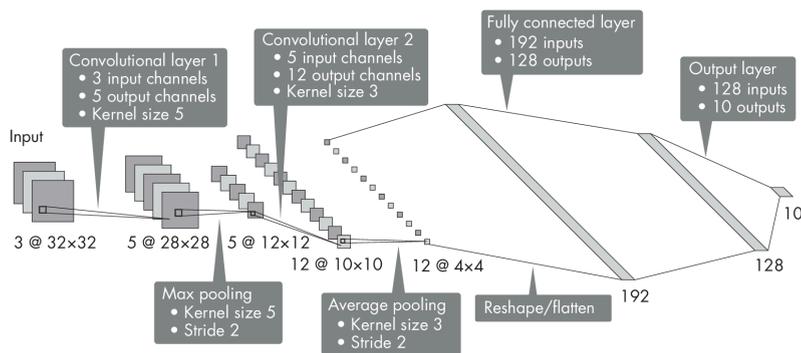


Figure 11.1

What is the number of trainable parameters in this convolutional network? We can approach this problem from left to right, computing the number of parameters for each layer and then summing up these counts to obtain the total number of parameters. Each layer's number of trainable parameters consists of weights and bias units.

Convolutional Layers

In a **convolutional layer**, the number of weights depends on the kernel's width and height and the number of input and output channels. The number of bias units depends on the number of output channels only. To illustrate the computation step by step, suppose we have a kernel width and height of 5, one input channel, and one output channel, as illustrated in Figure 11.2.

Tips: 卷积层的参数数量，取决于 **卷积核** (kernel) 的宽度、高度、输入通道数和输出通道数。

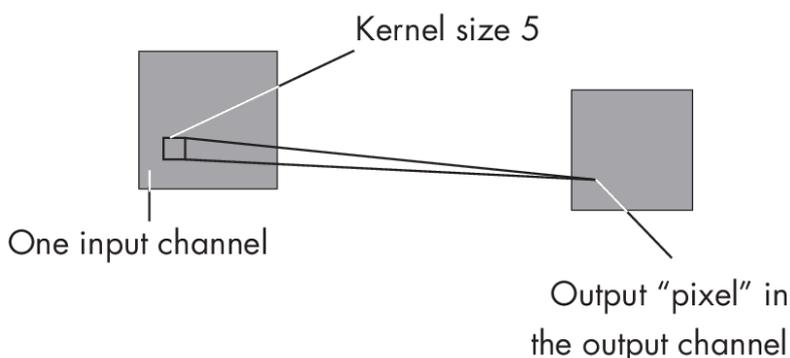


Figure 11.2



In this case, we have 26 parameters, since we have $5 \times 5 = 25$ weights via the kernel plus the bias unit. The computation to determine an output value or pixel z is $z = b + \sum_j w_j x_j$, where x_j represents an input pixel, w_j represents a weight parameter of the kernel, and b is the bias unit.

Tips: 一个卷积核 **kernel** 的参数量, **weights** = 宽度 x 高度。

Now, suppose we have three input channels, as illustrated in Figure 11.3.

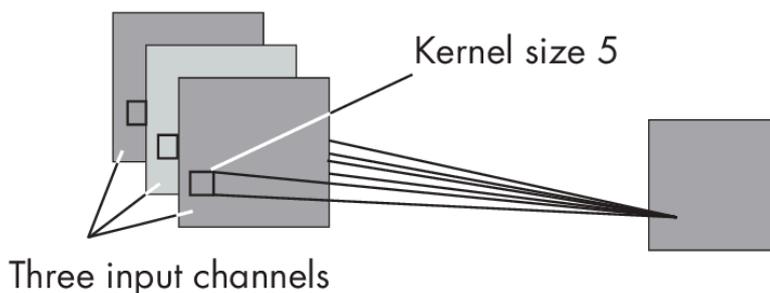


Figure 11.3

In that case, we compute the output value by performing the aforementioned operation, $\sum_j w_j x_j$, for each input channel and then add the bias unit. For three input channels, this would involve three different kernels with three sets of weights:

$$z = \sum_j w_j^{(1)} x_j + \sum_j w_j^{(2)} x_j + \sum_j w_j^{(3)} x_j + b$$

Since we have three sets of weights ($w^{(1)}$, $w^{(2)}$, and $w^{(3)}$ for $j = [1, 25]$), we have $3 \times 25 + 1 = 76$ parameters in this convolutional layer.

Tips: 每个输入通道, 对应的卷积核 **kernel**, 都是独立的参数。

We use one kernel for each output channel, where each kernel is unique to a given output channel. Thus, if we extend the number of output channels from one to five, as shown in Figure 11.4, we extend the number of parameters by a factor of 5.

In other words, if the kernel for one output channel has 76 parameters, the 5 kernels required for the five output channels will have $5 \times 76 = 380$ parameters.

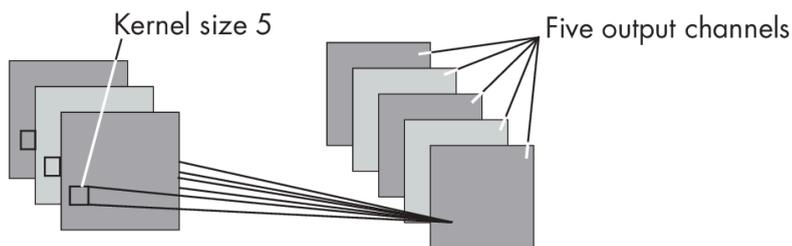


Figure 11.4

Returning to the neural network architecture illustrated in Figure [11.1](#) at the beginning of this section, we compute the number of parameters in the convolutional layers based on the kernel size and number of input and output channels. For example, the first convolutional layer has three input channels, five output channels, and a kernel size of 5. Thus, its number of parameters is $5 \times (5 \times 5 \times 3) + 5 = 380$.

The second convolutional layer, with five input channels, 12 output channels, and a kernel size of 3, has $12 \times (3 \times 3 \times 5) + 12 = 552$ parameters.

Since the **pooling layers** do not have any trainable parameters, we can count $380 + 552 = 932$ for the convolutional part of this architecture.

Next, let's see how we can compute the number of parameters of fully connected layers.

Fully Connected Layers

Counting the number of parameters in a **fully connected layer** is relatively straightforward. A fully connected node connects each input node to each output node, so the number of weights is the number of inputs times the number of outputs plus the bias units added to the output. For example, if we have a fully connected layer with five inputs and three outputs, as shown in Figure [11.5](#), we have $5 \times 3 = 15$ weights and three bias units, that is, 18 parameters total.

Tips: 一个全连接层 **fully connected layer** 的参数量, **weights** = 输入节点数 x 输出节点数。

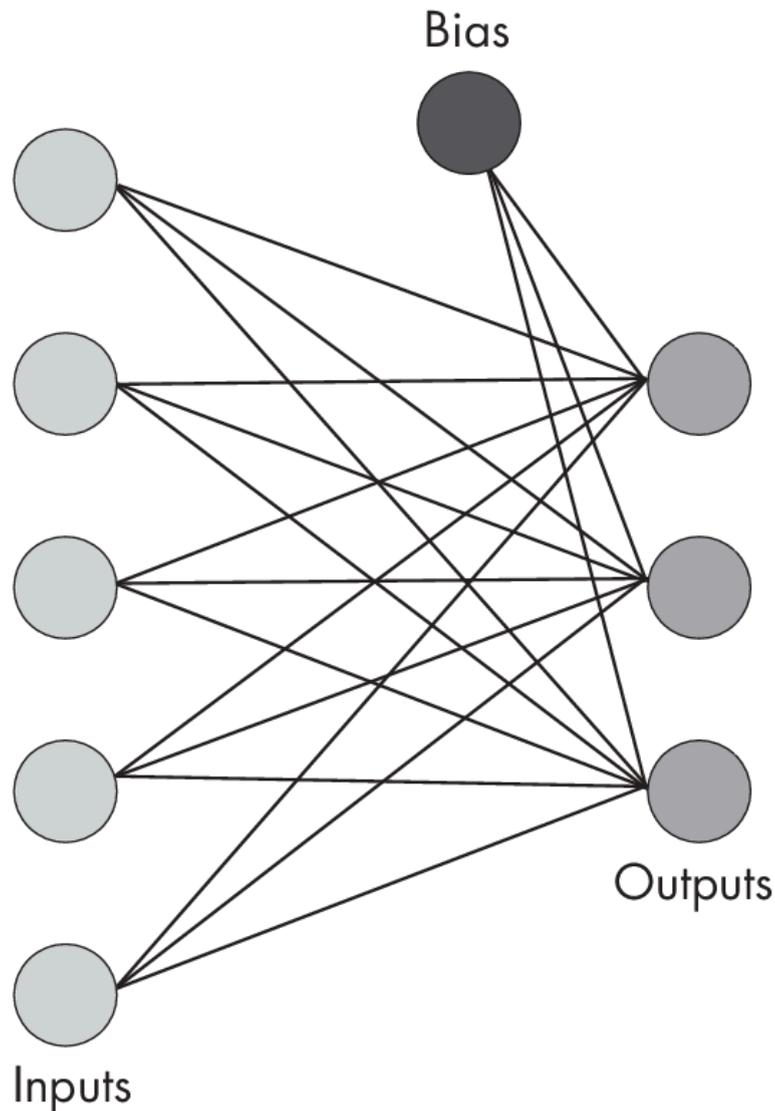


Figure 11.5

Returning once more to the neural network architecture illustrated in Figure [11.1](#),

we can now calculate the parameters in the fully connected layers as follows:

$192 \times 128 + 128 = 24,704$ in the first fully connected layer and $128 \times 10 + 10 = 1,290$ in the second fully connected layer, the output layer.

Hence, we have $24,704 + 1,290 = 25,994$ in the fully connected part of this network.

After adding the 932 parameters from the convolutional layers and the 25,994 parameters from the fully connected layers, we can conclude that this network's total number of parameters is 26,926.

As a bonus, interested readers can find PyTorch code to compute the number of parameters programmatically in the *supplementary/q11-conv-size* subfolder at

<https://github.com/rasbt/MachineLearning-QandAI-book>.



Practical Applications

Why do we care about the number of parameters at all? First, we can use this number to estimate a model's complexity. As a rule of thumb, the more parameters there are, the more training data we'll need to train the model well.

Tips: 模型 **参数的数量**，是衡量模型 **复杂度** 的重要指标，用于估算所需的 **训练数据量**。

The number of parameters also lets us estimate the size of the neural network, which in turn helps us estimate whether the network can fit into GPU memory. Although the memory requirement during training often exceeds the model size due to the additional memory required for carrying out matrix multiplications and storing gradients, model size gives us a ballpark sense of whether training the model on a given hardware setup is feasible.

Tips: 模型参数的数量，是衡量模型 **大小** 的重要指标，用于估算模型是否能 **fit** 到 **GPU** 的 **内存** 中。

Exercises

11-1. Suppose we want to optimize the neural network using a plain stochastic gradient descent (SGD) optimizer or the popular Adam optimizer. What are the respective numbers of parameters that need to be stored for SGD and Adam?

11-2. Suppose we're adding three batch normalization (BatchNorm) layers: one after the first convolutional layer, one after the second convolutional layer, and another one after the first fully connected layer (we typically do not want to add BatchNorm layers to the output layer). How many additional parameters do these three BatchNorm layers add to the model?



Chapter 12: Fully Connected and Convolutional Layers

Under which circumstances can we replace fully connected layers with convolutional layers to perform the same computation?

Replacing fully connected layers with convolutional layers can offer advantages in terms of hardware optimization, such as by utilizing specialized hardware accelerators for convolution operations. This can be particularly relevant for edge devices.

Tips: **卷积层** 替代 **全连接层**，有下面收益

- 卷积操作可以 **硬件加速**
- 这在 **边缘设备** 上非常关键。

There are exactly two scenarios in which fully connected layers and convolutional layers are equivalent: when the size of the convolutional filter is equal to the size of the receptive field and when the size of the convolutional filter is 1. As an illustration of these two scenarios, consider a fully connected layer with two input and four output units, as shown in Figure [12.1](#).

Tips: **全连接层** 和 **卷积层** 在两种情况下是 **等价** 的：

- 当 **卷积核** 的大小等于 **感受野** 的大小。
- 当 **卷积核** 的大小为 **1**。

receptive field 感受野，在 CNN 和 RNN 中，有不同的含义。

- 在卷积神经网络 (CNN) 中，感受野是指网络中某个特定层的神经元在输入图像上映射的区域大小（像素范围）。它是 **空间** 维度上的概念。
- 在循环神经网络 (RNN) 中，感受野是 **时间** 维度上的概念。它衡量的是当前状态在时间轴上向后能追溯到多远的输入信息。

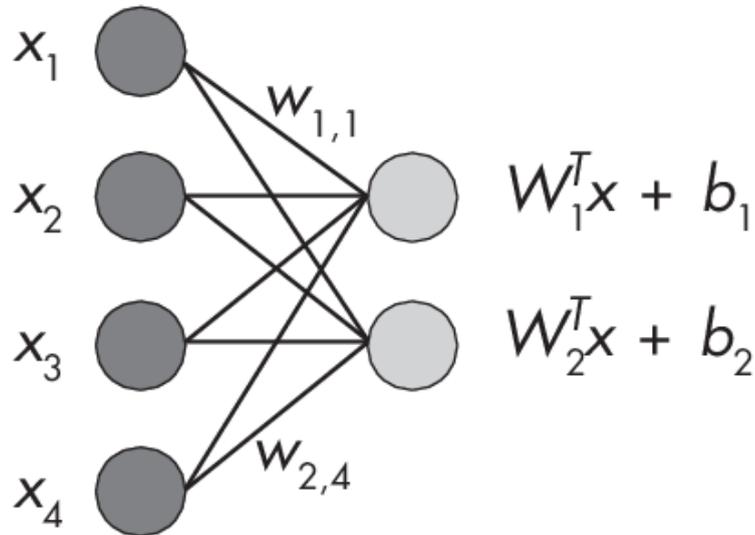


Figure 12.1

The fully connected layer in this figure consists of eight weights and two bias units. We can compute the output nodes via the following dot products:

Node 1

$$w_{1,1} \times x_1 + w_{1,2} \times x_2 + w_{1,3} \times x_3 + w_{1,4} \times x_4 + b_1$$

Node 2

$$w_{2,1} \times x_1 + w_{2,2} \times x_2 + w_{2,3} \times x_3 + w_{2,4} \times x_4 + b_2$$

The following two sections illustrate scenarios in which convolutional layers can be defined to produce exactly the same computation as the fully connected layer described.

When the Kernel and Input Sizes Are Equal

Let's start with the first scenario, where the size of the convolutional filter is equal to the size of the receptive field. Recall from Chapter [\[ch11\]](#) how we compute a number of parameters in a convolutional kernel with one input channel and multiple output channels. We have a kernel size of 2×2 , one input channel, and two output channels. The input size is also 2×2 , a reshaped version of the four inputs depicted in Figure [12.2](#).

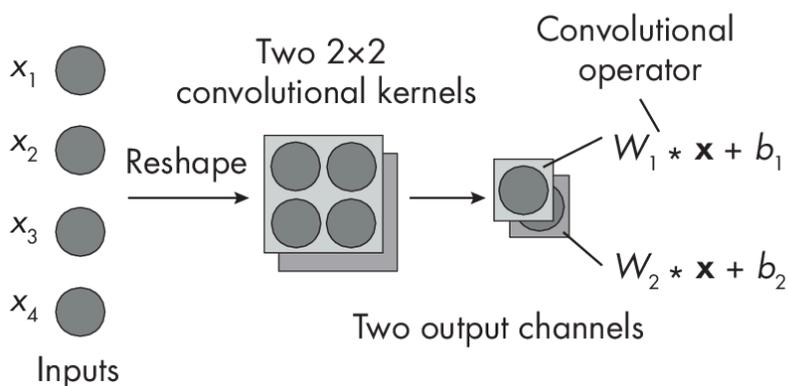


Figure 12.2

If the convolutional kernel dimensions equal the input size, as depicted in Figure 12.2, there is no sliding window mechanism in the convolutional layer. For the first output channel, we have the following set of weights:

$$W_1 = \begin{bmatrix} w_{1,1} & w_{1,2} \\ w_{1,3} & w_{1,4} \end{bmatrix}$$

For the second output channel, we have the following set of weights:

$$W_2 = \begin{bmatrix} w_{2,1} & w_{2,2} \\ w_{2,3} & w_{2,4} \end{bmatrix}$$

If the inputs are organized as

$$\mathbf{x} = \begin{bmatrix} x_1 & x_2 \\ x_3 & x_4 \end{bmatrix}$$

we calculate the first output channel as $o_1 = \sum_i (W_1 \times x_i) + b_1$, where the convolutional operator $*$ is equal to an element-wise multiplication. In other words, we perform an element-wise multiplication between two matrices, W_1 and \mathbf{x} , and then compute the output as the sum over these elements; this equals the dot product in the fully connected layer. Lastly, we add the bias unit. The computation for the second output channel works analogously: $o_2 = \sum_i (W_2 \times x_i) + b_2$.

As a bonus, the supplementary materials for this book include PyTorch code to show this equivalence with a hands-on example in the [supplementary/q12-fc-cnn-equivalence](https://github.com/rasbt/MachineLearning-QandAI-book) subfolder at <https://github.com/rasbt/MachineLearning-QandAI-book>.



When the Kernel Size Is 1

The second scenario assumes that we reshape the input into an input "image" with 1×1 dimensions where the number of "color channels" equals the number of input features, as depicted in Figure 12.3.

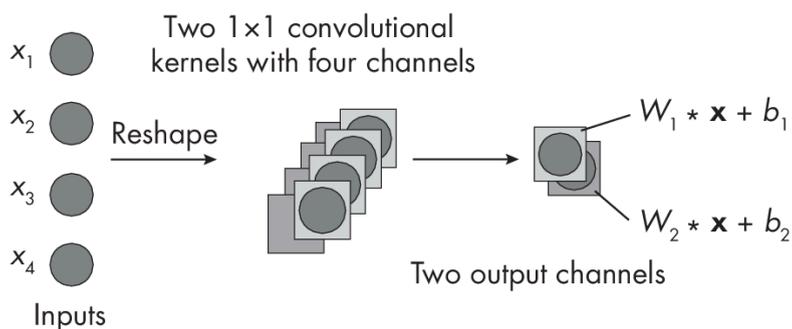


Figure 12.3

Each kernel consists of a stack of weights equal to the number of input channels. For instance, for the first output layer, the weights are

$$W_1 = \begin{bmatrix} w_1^{(1)} & w_1^{(2)} & w_1^{(3)} & w_1^{(4)} \end{bmatrix}$$

while the weights for the second channel are:

$$W_2 = \begin{bmatrix} w_2^{(1)} & w_2^{(2)} & w_2^{(3)} & w_2^{(4)} \end{bmatrix}$$

To get a better intuitive understanding of this computation, check out the illustrations in Chapter [ch11], which describe how to compute the parameters in a convolutional layer.

Recommendations

The fact that fully connected layers can be implemented as equivalent convolutional layers does not have immediate performance or other advantages on standard computers. However, replacing fully connected layers with convolutional layers can offer advantages in combination with developing specialized hardware accelerators for convolution operations.

Moreover, understanding the scenarios where fully connected layers are equivalent to convolutional layers aids in understanding the mechanics of these

layers. It also lets us implement convolutional neural networks without any use of fully connected layers, if desired, to simplify code implementations.



Tips: 进一步，理解 **卷积层** 和 **全连接层** 的等价性，有助于理解这些层的机制。

此外，如果需要，我们可以实现卷积神经网络，而不使用任何全连接层，以简化代码实现。

Exercises

12-1. How would increasing the stride affect the equivalence discussed in this chapter?

12-2. Does padding affect the equivalence between fully connected layers and convolutional layers?



Chapter 13: Large Training Sets for Vision Transformers

Why do vision transformers (ViTs) generally require larger training sets than convolutional neural networks (CNNs)?

Each machine learning algorithm and model encodes a particular set of assumptions or prior knowledge, commonly referred to as *inductive biases*, in its design. Some inductive biases are workarounds to make algorithms computationally more feasible, other inductive biases are based on domain knowledge, and some inductive biases are both.

CNNs and ViTs can be used for the same tasks, including image classification, object detection, and image segmentation. CNNs are mainly composed of convolutional layers, while ViTs consist primarily of multi-head attention blocks (discussed in Chapter [\[ch08\]](#) in the context of transformers for natural language inputs).

CNNs have more inductive biases that are hardcoded as part of the algorithmic design, so they generally require less training data than ViTs. In a sense, ViTs are given more degrees of freedom and can or must learn certain inductive biases from the data (assuming that these biases are conducive to optimizing the training objective). However, everything that needs to be learned requires more training examples.

The following sections explain the main inductive biases encountered in CNNs and how ViTs work well without them.

Inductive Biases in CNNs

The following are the primary inductive biases that largely define how CNNs function:

Local connectivity In CNNs, each unit in a hidden layer is connected to only a subset of neurons in the previous layer. We can justify this restriction by assuming that neighboring pixels are more relevant to each other than pixels that are



farther apart. As an intuitive example, consider how this assumption applies to the context of recognizing edges or contours in an image.

Weight sharing Via the convolutional layers, we use the same small set of weights (the kernels or filters) throughout the whole image. This reflects the assumption that the same filters are useful for detecting the same patterns in different parts of the image.

Hierarchical processing CNNs consist of multiple convolutional layers to extract features from the input image. As the network progresses from the input to the output layers, low-level features are successively combined to form increasingly complex features, ultimately leading to the recognition of more complex objects and shapes. Furthermore, the convolutional filters in these layers learn to detect specific patterns and features at different levels of abstraction.

Spatial invariance CNNs exhibit the mathematical property of spatial invariance, meaning the output of a model remains consistent even if the input signal is shifted to a different location within the spatial domain. This characteristic arises from the combination of local connectivity, weight sharing, and the hierarchical architecture mentioned earlier.

The combination of local connectivity, weight sharing, and hierarchical processing in a CNN leads to spatial invariance, allowing the model to recognize the same pattern or feature regardless of its location in the input image.

Translation invariance is a specific case of spatial invariance in which the output remains the same after a shift or translation of the input signal in the spatial domain. In this context, the emphasis is solely on moving an object to a different location within an image without any rotations or alterations of its other attributes.

In reality, convolutional layers and networks are not truly translation-invariant; rather, they achieve a certain level of translation equivariance. What is the difference between translation invariance and equivariance? *Translation invariance* means that the output does not change with an input shift, while *translation equivariance* implies that the output shifts with the input in a corresponding manner. In other words, if we shift the input object to the right, the results will correspondingly shift to the right, as illustrated in Figure [13.1](#).

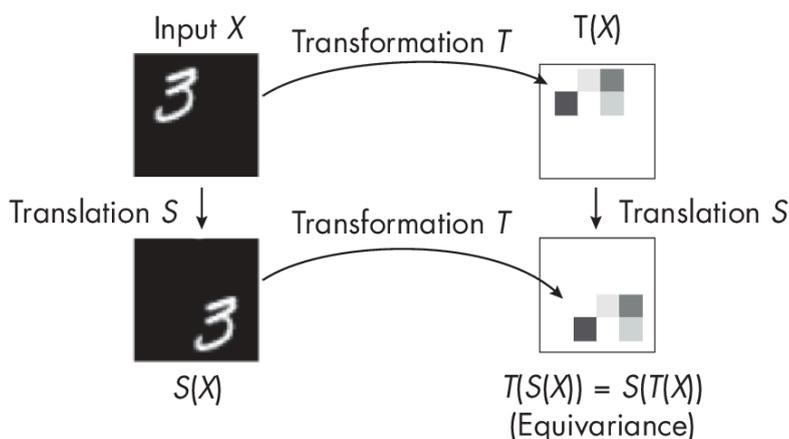


Figure 13.1

As Figure 13.1 shows, under translation invariance, we get the same output pattern regardless of the order in which we apply the operations: transformation followed by translation or translation followed by transformation.

As mentioned earlier, CNNs achieve translation equivariance through a combination of their local connectivity, weight sharing, and hierarchical processing properties. Figure 13.2 depicts a convolutional operation to illustrate the local connectivity and weight-sharing priors. This figure demonstrates the concept of translation equivariance in CNNs, in which a convolutional filter captures the input signal (the two dark blocks) irrespective of where it is located in the input.

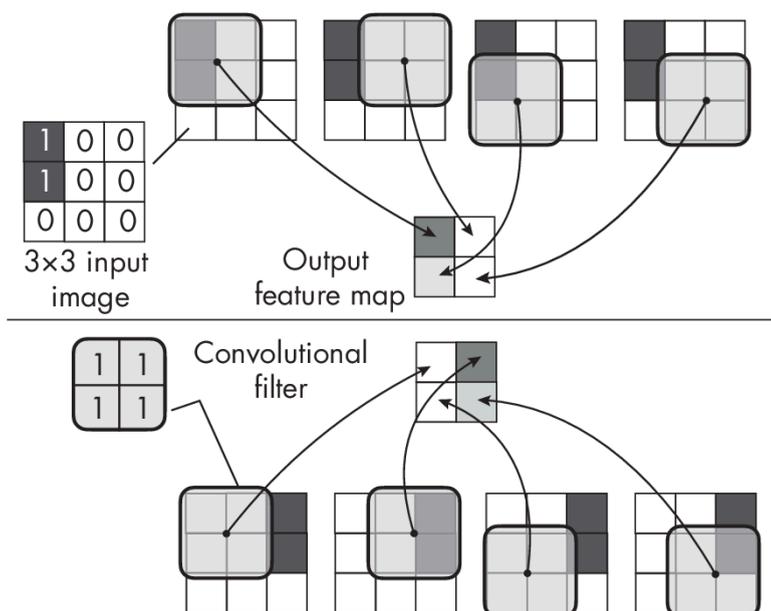


Figure 13.2



Figure 13.2 shows a 3×3 input image that consists of two nonzero pixel values in the upper-left corner (top portion of the figure) or upper-right corner (bottom portion of the figure). If we apply a 2×2 convolutional filter to these two input image scenarios, we can see that the output feature maps contain the same extracted pattern, which is on either the left (top of the figure) or the right (bottom of the figure), demonstrating the translation equivariance of the convolutional operation.

For comparison, a fully connected network such as a multilayer perceptron lacks this spatial invariance or equivariance. To illustrate this point, picture a multilayer perceptron with one hidden layer. Each pixel in the input image is connected with each value in the resulting output. If we shift the input by one or more pixels, a different set of weights will be activated, as illustrated in Figure 13.3.

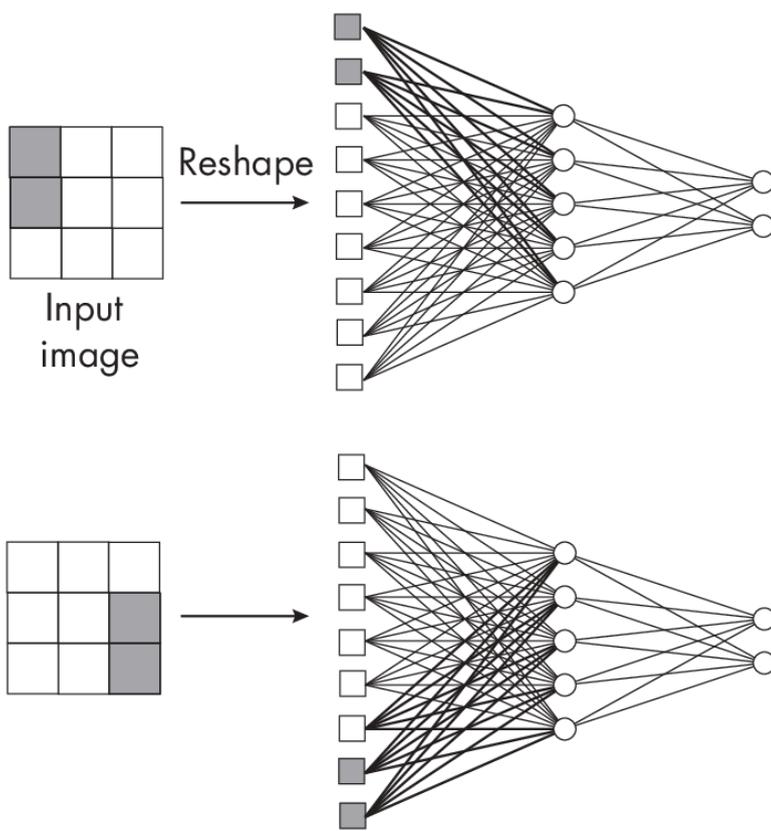


Figure 13.3

Like fully connected networks, ViT architecture (and transformer architecture in general) lacks the inductive bias for spatial invariance or equivariance. For instance, the model produces different outputs if we place the same object in two different spatial locations within an image. This is not ideal, as the semantic meaning of an object (the concept that an object represents or conveys) remains the same based on its location. Consequently, it must learn these invariances



directly from the data. To facilitate learning useful patterns present in CNNs requires pretraining over a larger dataset.

A common workaround for adding positional information in ViTs is to use relative positional embeddings (also known as *relative positional encodings*) that consider the relative distance between two tokens in the input sequence. However, while relative embeddings encode information that helps transformers keep track of the relative location of tokens, the transformer still needs to learn from the data whether and how far spatial information is relevant for the task at hand.

ViTs Can Outperform CNNs

The hardcoded assumptions via the inductive biases discussed in previous sections reduce the number of parameters in CNNs substantially compared to fully connected layers. On the other hand, ViTs tend to have larger numbers of parameters than CNNs, which require more training data. (Refer to Chapter [\[ch11\]](#) for a refresher on how to precisely calculate the number of parameters in fully connected and convolutional layers.)

ViTs may underperform compared to popular CNN architectures without extensive retraining, but they can perform very well with a sufficiently large pretraining dataset. In contrast to language transformers, where unsupervised pretraining (such as self-supervised learning, discussed in Chapter [\[ch02\]](#)) is a preferred choice, vision transformers are often pretrained using large, labeled datasets like ImageNet, which provides millions of labeled images for training, and regular supervised learning.

An example of ViTs surpassing the predictive performance of CNNs, given enough data, can be observed from initial research on the ViT architecture, as shown in the paper "An Image Is Worth 16x16 Words: Transformers for Image Recognition at Scale."? This study compared ResNet, a type of convolutional network, with the original ViT design using different dataset sizes for pretraining. The findings also showed that the ViT model excelled over the convolutional approach only after being pretrained on a minimum of 100 million images.

Inductive Biases in ViTs

ViTs also possess some inductive biases. For example, vision transformers *patchify* the input image to process each input patch individually. Here, each patch can attend to all other patches so that the model learns relationships between far-apart patches in the input image, as illustrated in Figure 13.4.

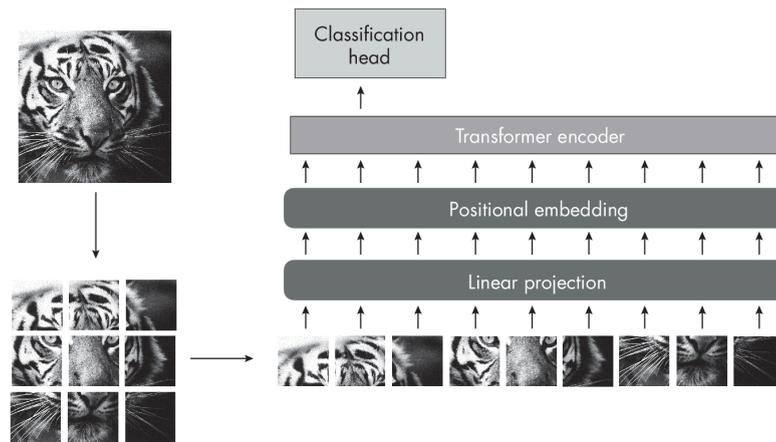


Figure 13.4

The patchify inductive bias allows ViTs to scale to larger image sizes without increasing the number of parameters in the model, which can be computationally expensive. By processing smaller patches individually, ViTs can efficiently capture spatial relationships between image regions while benefiting from the global context captured by the self-attention mechanism.

This raises another question: how and what do ViTs learn from the training data? ViTs learn more uniform feature representations across all layers, with self-attention mechanisms enabling early aggregation of global information. In addition, the residual connections in ViTs strongly propagate features from lower to higher layers, in contrast to the more hierarchical structure of CNNs.

ViTs tend to focus more on global than local relationships because their self-attention mechanism allows the model to consider long-range dependencies between different parts of the input image. Consequently, the self-attention layers in ViTs are often considered low-pass filters that focus more on shapes and curvature.

In contrast, the convolutional layers in CNNs are often considered high-pass filters that focus more on texture. However, keep in mind that convolutional layers can act as both high-pass and low-pass filters, depending on the learned filters at each layer. High-pass filters detect an image's edges, fine details, and texture, while low-pass filters capture more global, smooth features and shapes. CNNs



achieve this by applying convolutional kernels of varying sizes and learning different filters at each layer.

Recommendations

ViTs have recently begun outperforming CNNs if enough data is available for pretraining. However, this doesn't make CNNs obsolete, as methods such as the popular EfficientNetV2 CNN architecture are less memory and data hungry.

Moreover, recent ViT architectures don't rely solely on large datasets, parameter numbers, and self-attention. Instead, they have taken inspiration from CNNs and added soft convolutional inductive biases or even complete convolutional layers to get the best of both worlds.

In short, vision transformer architectures without convolutional layers generally have fewer spatial and locality inductive biases than convolutional neural networks. Consequently, vision transformers need to learn data-related concepts such as local relationships among pixels. Thus, vision transformers require more training data to achieve good predictive performance and produce acceptable visual representations in generative modeling contexts.

Exercises

13-1. Consider the patchification of the input images shown in Figure [13.4](#). The size of the resulting patches controls a computational and predictive performance trade-off. The optimal patch size depends on the application and desired trade-off between computational cost and model performance. Do smaller patches typically result in higher or lower computational costs?

13-2. Following up on the previous question, do smaller patches typically lead to a higher or lower prediction accuracy?

References

- The paper proposing the original vision transformer model: Alexey Dosovitskiy et al., "An Image Is Worth 16x16 Words: Transformers for Image Recognition at



Scale"? (2020), <https://arxiv.org/abs/2010.11929>.

- A workaround for adding positional information in ViTs is to use relative positional embeddings: Peter Shaw, Jakob Uszkoreit, and Ashish Vaswani, "Self-Attention with Relative Position Representations"? (2018), <https://arxiv.org/abs/1803.02155>.
 - Residual connections in ViTs strongly propagate features from lower to higher layers, in contrast to the more hierarchical structure of CNNs: Maithra Raghu et al., "Do Vision Transformers See Like Convolutional Neural Networks?" (2021), <https://arxiv.org/abs/2108.08810>.
 - A detailed research article covering the EfficientNetV2 CNN architecture: Mingxing Tan and Quoc V. Le, "Smaller Models and Faster Training"? (2021), <https://arxiv.org/abs/2104.00298>.
 - A ViT architecture that also incorporates convolutional layers: Stéphane d'Ascoli et al., "ConViT: Improving Vision Transformers with Soft Convolutional Inductive Biases"? (2021), <https://arxiv.org/abs/2103.10697>.
 - Another example of a ViT using convolutional layers: Haiping Wu et al., "CvT: Introducing Convolutions to Vision Transformers"? (2021), <https://arxiv.org/abs/2103.15808>.
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Chapter 14: The Distributional Hypothesis

What is the **distributional hypothesis** in natural language processing (NLP)? Where is it used, and how far does it hold true?

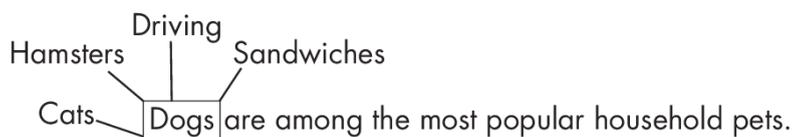
The **distributional hypothesis** is a linguistic theory suggesting that words occurring in the same contexts tend to have similar meanings, according to the original source, "Distributional Structure" by Zellig S. Harris. Succinctly, the more similar the meanings of two words are, the more often they appear in similar contexts.

Tips: 分布假设 (distributional hypothesis), 也称为分布语义学 (distributional semantics), 用于描述单词在上下文中的分布模式。它认为, 在 **相似的上下文** 中出现的 **单词** 往往具有 **相似的含义**。

Consider the sentence in Figure 14.1, for example. The words *cats* and *dogs* often occur in similar contexts, and we could replace *cats* with *dogs* without making the sentence sound awkward. We could also replace *cats* with *hamsters*, since both are mammals and pets, and the sentence would still sound plausible. However, replacing *cats* with an unrelated word such as *sandwiches* would render the sentence clearly wrong, and replacing *cats* with the unrelated word *driving* would also make the sentence grammatically incorrect.

Tips: 图1.1中的句子。*cats*和*dogs*经常出现在相似的上下文中, 我们可以将*cats*替换为*dogs*, 而不会让句子听起来奇怪。

- 我们也可以将*cats*替换为*hamsters*, 因为它们都是哺乳动物和宠物, 句子听起来仍然合理。
- 但是, 如果将*cats*替换为不相关的单词*sandwiches*, 句子会变得明显错误,
- 如果将*cats*替换为不相关的单词*driving*, 句子也会变得语法错误。



**Figure 14.1**

It is easy to construct counterexamples using polysemous words, that is, words that have multiple meanings that are related but not identical. For example, consider the word *bank*. As a noun, it can refer to a financial institution, the "rising ground bordering a river," the "steep incline of a hill," or a "protective cushioning rim" (according to the Merriam-Webster dictionary). It can even be a verb: to bank on something means to rely or depend on it. These different meanings have different distributional properties and may not always occur in similar contexts.

Nonetheless, the distributional hypothesis is quite useful. Word embeddings (introduced in Chapter [ch01]) such as Word2vec, as well as many large language transformer models, rely on this idea. This includes the masked language model in BERT and the next-word pretraining task used in GPT.

Tips: 尽管存在反例，分布假设在实际应用中非常有用。

- Word2vec 等词嵌入 (word embeddings) 模型以及许多大型语言模型 (large language models) 都基于这个概念。
- 这包括 BERT 中的掩码语言模型，和 GPT 中的下一个词预训练任务。

Word2vec, BERT, and GPT

The Word2vec approach uses a simple, two-layer neuralnetwork to encode words into embedding vectors such that the embedding vectors of similar words are both semantically and syntactically close. There are two ways to train a Word2vec model: the continuous bag-of-words (CBOW) approach and the skip-gram approach. When using CBOW, the Word2vec model learns to predict the current words by using the surrounding context words. Conversely, in the skip-gram model, Word2vec predicts the context words from a selected word. While skip-gram is more effective for infrequent words, CBOW is usually faster to train.

Tips: FIXME 跳字模型 没理解???

- Word2vec 是一种使用简单两层神经网络将单词编码为嵌入向量的方法。



- 有两种训练Word2vec模型的方法：**连续词袋**（CBOW）方法和**跳字**（skip-gram）方法。
- 当使用CBOW时，Word2vec模型学习通过使用周围上下文单词来预测当前单词。
- 相反，在跳字模型中，Word2vec从选定的单词预测上下文单词。
- 尽管跳字模型对于不常见的单词更有效，但CBOW通常训练速度更快。

After training, word embeddings are placed within the vector space so that words with common contexts in the corpus--that is, words with semantic and syntactic similarities--are positioned close to each other, as illustrated in Figure 14.2. Conversely, dissimilar words are located farther apart in the embedding space.

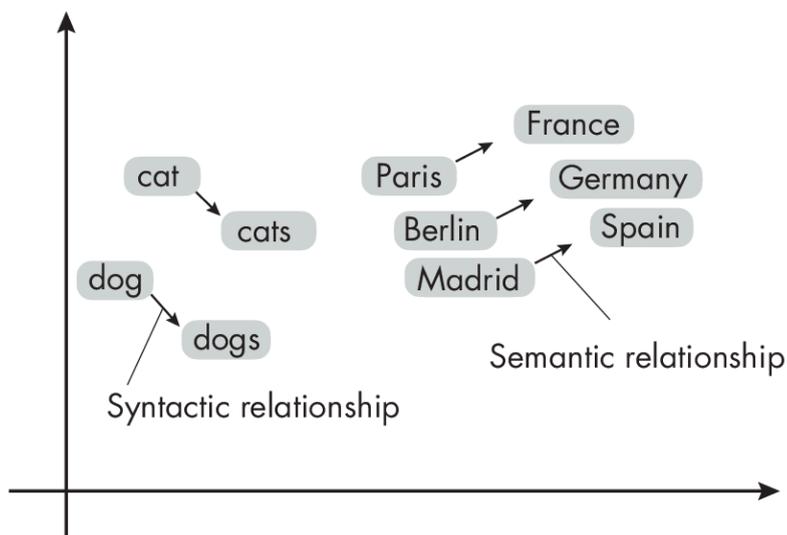


Figure 14.2

BERT is an LLM based on the transformer architecture (see Chapter [ch08]) that uses a masked language modeling approach that involves masking (hiding) some of the words in a sentence. Its task is to predict these masked words based on the other words in the sequence, as illustrated in Figure 14.3. This is a form of the self-supervised learning used to pretrain LLMs (see Chapter [ch02] for more on self-supervised learning). The pretrained model produces embeddings in which similar words (or tokens) are close in the embedding space.

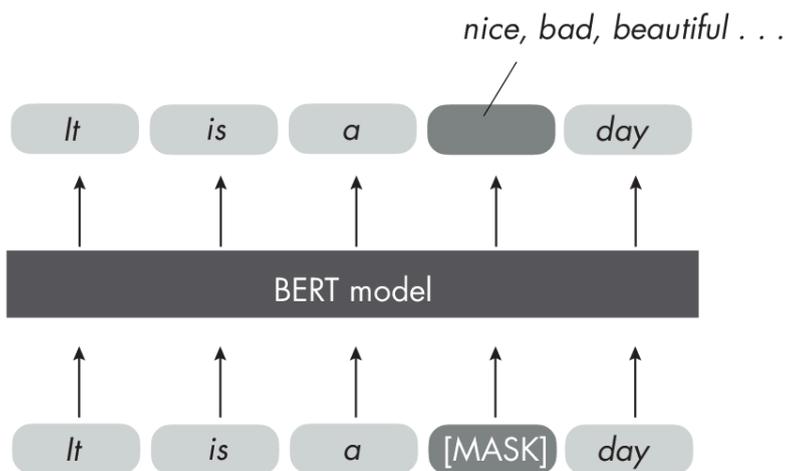


Figure 14.3

GPT, which like BERT is also an LLM based on the transformer architecture, functions as a decoder. Decoder-style models like GPT learn to predict subsequent words in a sequence based on the preceding ones, as illustrated in Figure 14.4. GPT contrasts with BERT, an encoder model, as it emphasizes predicting what follows rather than encoding the entire sequence simultaneously.

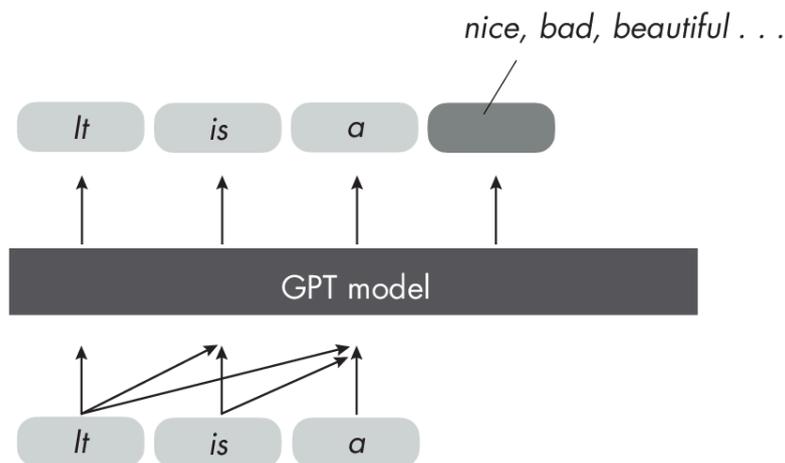


Figure 14.4

Where BERT is a bidirectional language model that considers the whole input sequence, GPT only strictly parses previous sequence elements. This means BERT is usually better suited for classification tasks, whereas GPT is more suited for text generation tasks. Similar to BERT, GPT produces high-quality contextualized word embeddings that capture semantic similarity.

Tips:

- **BERT** 是一种 **双向** 语言模型，考虑 **整个输入序列**。

- **GPT** 只严格解析 **前一个序列元素**。
- 这意味着 **BERT** 通常更适合分类任务，而 **GPT** 更适合文本生成任务。
- 与 **BERT** 类似，**GPT** 产生高质量的 **上下文单词嵌入**，捕捉语义相似性。



Does the Hypothesis Hold?

For large datasets, the distributional hypothesis more or less holds true, making it quite useful for understanding and modeling language patterns, word relationships, and semantic meanings. For example, this concept enables techniques like word embedding and semantic analysis, which, in turn, facilitate natural language processing tasks such as text classification, sentiment analysis, and machine translation.

Tips:

- 对于大型数据集，**分布假设**或多或少是正确的，对于理解语言模式、单词关系和语义含义非常有用。
- 例如，这个概念启用了像 **词嵌入** 和 **语义分析** 这样的技术，这些技术反过来又促进了自然语言处理任务，如 **文本分类**、**情感分析** 和 **机器翻译**。

In conclusion, while there are counterexamples in which the distributional hypothesis does not hold, it is a very useful concept that forms the cornerstone of modern language transformer models.

Exercises

14-1. Does the distributional hypothesis hold true in the case of homophones, or words that sound the same but have different meanings, such as *there* and *their*?

14-2. Can you think of another domain where a concept similar to the distributional hypothesis applies? (Hint: think of other input modalities for neural networks.)



References

- The original source describing the distributional hypothesis: Zellig S. Harris, "Distributional Structure"? (1954), <https://doi.org/10.1080/00437956.1954.11659520>.
 - The paper introducing the Word2vec model: Tomas Mikolov et al., "Efficient Estimation of Word Representations in Vector Space"? (2013), <https://arxiv.org/abs/1301.3781>.
 - The paper introducing the BERT model: Jacob Devlin et al., "BERT: Pre-training of Deep Bidirectional Transformers for Language Understanding"? (2018), <https://arxiv.org/abs/1810.04805>.
 - The paper introducing the GPT model: Alec Radford and Karthik Narasimhan, "Improving Language Understanding by Generative Pre-Training"? (2018), <https://www.semanticscholar.org/paper/Improving-Language-Understanding-by-Generative-Radford-Narasimhan/cd18800a0fe0b668a1cc19f2ec95b5003d0a5035>.
 - BERT produces embeddings in which similar words (or tokens) are close in the embedding space: Nelson F. Liu et al., "Linguistic Knowledge and Transferability of Contextual Representations"? (2019), <https://arxiv.org/abs/1903.08855>.
 - The paper showing that GPT produces high-quality contextualized word embeddings that capture semantic similarity: Fabio Petroni et al., "Language Models as Knowledge Bases"? (2019), <https://arxiv.org/abs/1909.01066>.
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Chapter 15: Data Augmentation for Text

How is data augmentation useful, and what are the most common augmentation techniques for text data?

Data augmentation is useful for artificially increasing dataset sizes to improve model performance, such as by reducing the degree of overfitting, as discussed in Chapter [\[ch05\]](#). This includes techniques often used in computer vision models, like rotation, scaling, and flipping.

Tips: 数据增强（**Data Augmentation**）是一种通过增加训练数据量，来提高模型泛化能力的技术。

数据增强可以分为两类：

1. 基于**规则**的数据增强：通过规则生成新的数据，如同义词替换、单词删除、单词位置交换、句子打乱、噪声注入等。
2. 基于**模型**的数据增强：通过模型生成新的数据，如 **GAN**、**VAE** 等。

数据增强通常包括以下几种技术：

1. 同义词替换（**Synonym Replacement**）：依赖同义词典，替换同义词；建议设置替换频率和相似度阈值。
2. 单词删除（**Word Deletion**）：随机删除句子中的某些单词；建议设置删除率。
3. 单词位置交换（**Word Position Swapping**）：随机交换句子中某些单词的位置；建议设置交换率。
4. 句子打乱（**Sentence Shuffling**）：随机打乱句子中某些句子的顺序；建议设置打乱率。
5. 噪声注入（**Noise Injection**）：在句子中添加一些随机噪声；建议设置噪声率。
6. 回译（**Back Translation**）：将句子翻译成另一种语言，再翻译回来；建议设置翻译模型。
7. 合成数据（**Synthetic Data**）：使用LLM生成新的数据；建议设置生成模型。



Similarly, there are several techniques for augmenting text data. The most common include synonym replacement, word deletion, word position swapping, sentence shuffling, noise injection, back translation, and text generated by LLMs. This chapter discusses each of these, with optional code examples in the *supplementary/q15-text-augment* subfolder at

<https://github.com/rasbt/MachineLearning-QandAI-book>.

Synonym Replacement

In *synonym replacement*, we randomly choose words in a sentence -- often nouns, verbs, adjectives, and adverbs -- and replace them with synonyms. For example, we might begin with the sentence "The cat quickly jumped over the lazy dog," and then augment the sentence as follows: "The cat rapidly jumped over the idle dog."?

Synonym replacement can help the model learn that different words can have similar meanings, thereby improving its ability to understand and generate text. In practice, synonym replacement often relies on a thesaurus such as WordNet. However, using this technique requires care, as not all synonyms are interchangeable in all contexts. Most automatic text replacement tools have settings for adjusting replacement frequency and similarity thresholds. However, automatic synonym replacement is not perfect, and you might want to apply post-processing checks to filter out replacements that might not make sense.

Tips: 自动 **同义词替换** 的结果并不准确，建议筛除不合适的替换结果。

Word Deletion

Word deletion is another data augmentation technique to help models learn. Unlike synonym replacement, which alters the text by substituting words with their synonyms, word deletion involves removing certain words from the text to create new variants while trying to maintain the overall meaning of the sentence. For example, we might begin with the sentence "The cat quickly jumped over the lazy dog" and then remove the word *quickly*: "The cat jumped over the lazy dog."?



By randomly deleting words in the training data, we teach the model to make accurate predictions even when some information is missing. This can make the model more robust when encountering incomplete or noisy data in real-world scenarios. Also, by deleting nonessential words, we may teach the model to focus on key aspects of the text that are most relevant to the task at hand.

However, we must be careful not to remove critical words that may significantly alter a sentence's meaning. For example, it would be suboptimal to remove the word *cat* in the previous sentence: "The quickly jumped over the lazy dog."? We must also choose the deletion rate carefully to ensure that the text still makes sense after words have been removed. Typical deletion rates might range from 10 percent to 20 percent, but this is a general guideline and could vary significantly based on the specific use case.

Tips: 单词删除，有时候会删除掉一些重要的词语，导致句子不通顺，建议设置删除率。一般删除率在 10% 到 20% 之间，但具体需要根据具体任务调整。

Word Position Swapping

In *word position swapping*, also known as *word shuffling* or *permutation*, the positions of words in a sentence are swapped or rearranged to create new versions of the sentence. If we begin with "The cat quickly jumped over the lazy dog,"? we might swap the positions of some words to get the following: "Quickly the cat jumped the over lazy dog."?

While these sentences may sound grammatically incorrect or strange in English, they provide valuable training information for data augmentation because the model can still recognize the important words and their associations with each other. However, this method has its limitations. For example, shuffling words too much or in certain ways can drastically change the meaning of a sentence or make it completely nonsensical. Moreover, word shuffling may interfere with the model's learning process, as the positional relationships between certain words can be vital in these contexts.

Tips: 词序打乱，有助于模型关注词语本身及其关系，而非固定语序。但过度打乱可能导致语义丢失或句子无意义。某些任务中，词语的顺序很重要，

随意打乱会影响模型学习效果。



Sentence Shuffling

In *sentence shuffling*, entire sentences within a paragraph or a document are rearranged to create new versions of the input text. By shuffling sentences within a document, we expose the model to different arrangements of the same content, helping it learn to recognize thematic elements and key concepts rather than relying on specific sentence order. This promotes a more robust understanding of the document's overall topic or category. Consequently, this technique is particularly useful for tasks that deal with document-level analysis or paragraph-level understanding, such as document classification, topic modeling, or text summarization.

In contrast to the aforementioned word-based methods (word position swapping, word deletion, and synonym replacement), sentence shuffling maintains the internal structure of individual sentences. This avoids the problem of altering word choice or order such that sentences become grammatically incorrect or change meaning entirely.

Sentence shuffling is useful when the order of sentences is not crucial to the overall meaning of the text. Still, it may not work well if the sentences are logically or chronologically connected. For example, consider the following paragraph: "I went to the supermarket. Then I bought ingredients to make pizza. Afterward, I made some delicious pizza."? Reshuffling these sentences as follows disrupts the logical and temporal progression of the narrative: "Afterward, I made some delicious pizza. Then I bought ingredients to make pizza. I went to the supermarket."?

Noise Injection

Noise injection is an umbrella term for techniques used to alter text in various ways and create variation in the texts. It may refer either to the methods described in the previous sections or to character-level techniques such as inserting random letters, characters, or typos, as shown in the following examples:



Random character insertion "The cat qzquickly jumped over the lazy dog."?

(Inserted a *z* in the word *quickly*.)

Random character deletion "The cat quickl jumped over the lazy dog."? (Deleted *y*

from the word *quickly*.)

Typo introduction "The cat qickuly jumped over the lazy dog."? (Introduced a typo

in *quickly*, changing it to *qickuly*.)

These modifications are beneficial for tasks that involve spell-checking and text correction, but they can also help make the model more robust to imperfect inputs.

Tips: 随机字符插入、删除、错别字，有助于模型学习拼写和语法错误，但过度使用可能导致模型过拟合。

Back Translation

Back translation is one of the most widely used techniques to create variation in texts. Here, a sentence is first translated from the original language into one or more different languages, and then it is translated back into the original language. Translating back and forth often results in sentences that are semantically similar to the original sentence but have slight variations in structure, vocabulary, or grammar. This generates additional, diverse examples for training without altering the overall meaning.

For example, say we translate "The cat quickly jumped over the lazy dog"? into German. We might get "Die Katze sprang schnell Ã¼ber den faulen Hund."? We could then translate this German sentence back into English to get "The cat jumped quickly over the lazy dog."?

The degree to which a sentence changes through backtranslation depends on the languages used and the specifics of the machine translation model. In this example, the sentence remains very similar. However, in other cases or with other languages, you might see more significant changes in wording or sentence structure while maintaining the same overall meaning.



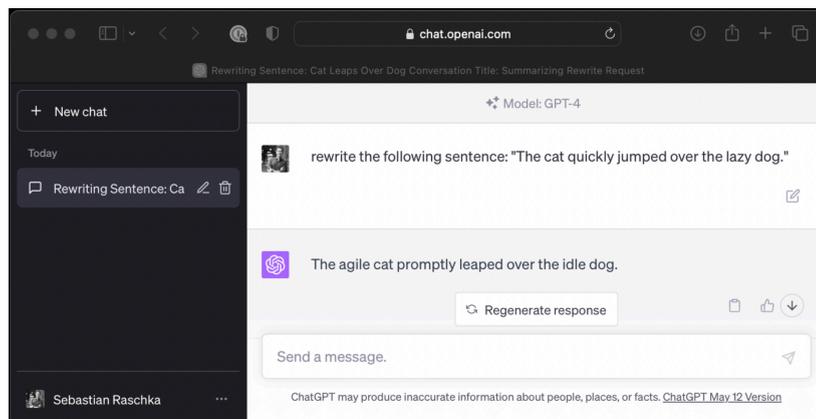
This method requires access to reliable machine translation models or services, and care must be taken to ensure that the back-translated sentences retain the essential meaning of the original sentences.

Tips: 回译，通过将句子翻译成另一种语言，再翻译回来，可以生成新的数据。但回译的结果有时并不准确，需要筛除不合适的回译结果。

Synthetic Data

Synthetic data generation is an umbrella term that describes methods and techniques used to create artificial data that mimics or replicates the structure of real-world data. All methods discussed in this chapter can be considered synthetic data generation techniques since they generate new data by making small changes to existing data, thus maintaining the overall meaning while creating something new.

Modern techniques to generate synthetic data now also include using decoder-style LLMs such as GPT (decoder-style LLMs are discussed in more detail in Chapter [\[ch17\]](#)). We can use these models to generate new data from scratch by using "complete the sentence"? or "generate example sentences"? prompts, among others. We can also use LLMs as alternatives to back translation, prompting them to rewrite sentences as shown in Figure [1.1](#).



Note that an LLM, as shown in Figure [1.1](#), runs in a nondeterministic mode by default, which means we can prompt it multiple times to obtain a variety of rewritten sentences.



Recommendations

The data augmentation techniques discussed in this chapter are commonly used in text classification, sentiment analysis, and other NLP tasks where the amount of available labeled data might be limited.

LLMs are usually pretrained on such a vast and diverse dataset that they may not rely on these augmentation techniques as extensively as in other, more specific NLP tasks. This is because LLMs aim to capture the statistical properties of the language, and the vast amount of data on which they are trained often provides a sufficient variety of contexts and expressions. However, in the fine-tuning stages of LLMs, where a pretrained model is adapted to a specific task with a smaller, task-specific dataset, data augmentation techniques might become more relevant again, mainly if the task-specific labeled dataset size is limited.

Tips: 数据增强 在LLM的 预训练 阶段可能 不太有用 ， 因为LLM已经在大规模数据上预训练过了。但在 微调阶段 ， 数据增强可能 更有用 ， 特别是当任务特定的标注数据集较小时。

Exercises

15-1. Can the use of text data augmentation help with privacy concerns?

15-2. What are some instances where data augmentation may not be beneficial for a specific task?

References

- The WordNet thesaurus: George A. Miller, "WordNet: A Lexical Database for English"? (1995), <https://dl.acm.org/doi/10.1145/219717.219748>.



Chapter 16: Self-Attention

Where does **self-attention** get its name, and how is it different from previously developed attention mechanisms?

Self-attention enables a neural network to refer to other portions of the input while focusing on a particular segment, essentially allowing each part the ability to "attend" to the whole input. The original attention mechanism developed for **recurrent neural networks** (RNNs) is applied between two different sequences: the encoder and the decoder embeddings. Since the attention mechanisms used in transformer-based large language models is designed to work on all elements of the same set, it is known as *self-attention*.

Tips: **self-attention** 允许神经网络在关注特定部分时，同时参考其他部分。每个部分都能“关注”整个输入。

This chapter first discusses an earlier attention mechanism developed for RNNs, the **Bahdanau** mechanism, in order to illustrate the motivation behind developing attention mechanism. We then compare the Bahdanau mechanism to the self-attention mechanism prevalent in transformer architectures today.

Attention in RNNs

One example of an attention mechanism used in **RNNs** to handle long sequences is *Bahdanau attention*. Bahdanau attention was developed to make machine learning models, particularly those used in translating languages, better at understanding long sentences. Before this type of attention, the whole input (such as a sentence in English) was squashed into a single chunk of information, and important details could get lost, especially if the sentence was long.

To understand the difference between regular attention and self-attention, let's begin with the illustration of the Bahdanau attention mechanism in Figure [16.1](#).

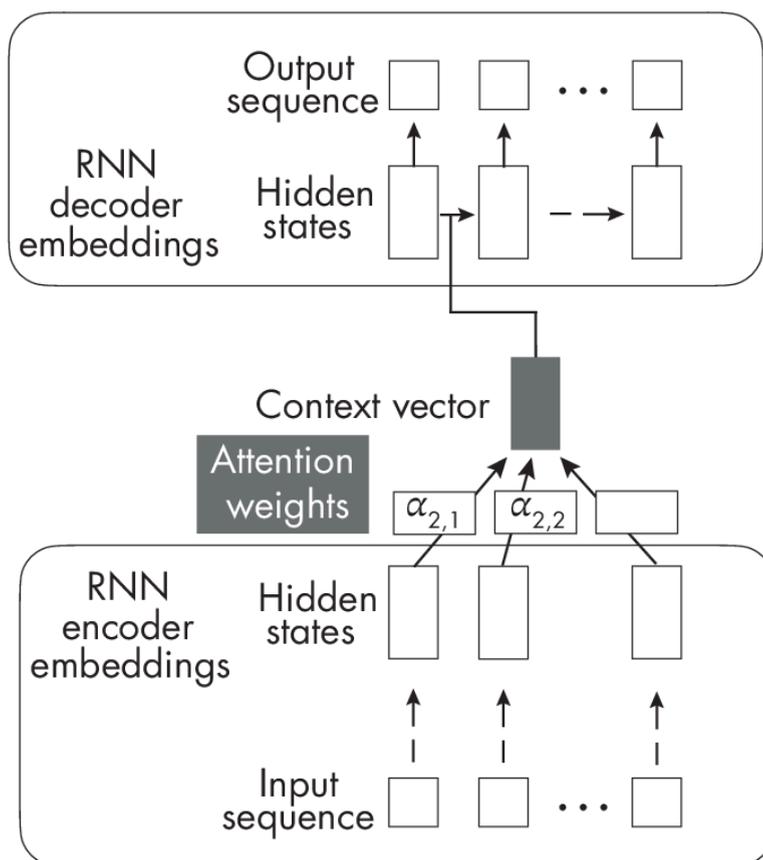


Figure 16.1

In Figure 16.1, the α values represent the attention weights for the second sequence element and each other element in the sequence from 1 to T . Furthermore, this original attention mechanism involves two RNNs .

- The RNN at the bottom, computing the attention weights, represents the encoder,
- while the RNN at the top, producing the output sequence, is a decoder.

Tips:

- 原始注意力机制是应用于两个不同序列的：编码器 和 解码器 嵌入。
- 对于每个生成的输出序列元素，解码器RNN 基于隐藏状态和编码器生成的上下文向量。
- 上下文向量 涉及输入序列的所有元素，是所有输入元素的加权和，其中，注意力分数 (α) 表示权重系数。
- 这允许 解码器 访问输入序列的所有元素（上下文）。



In short, the original attention mechanism developed for RNNs is applied between two different sequences: the encoder and decoder embeddings. For each generated output sequence element, the decoder RNN at the top is based on a hidden state plus a context vector generated by the encoder. This context vector involves *all* elements of the input sequence and is a weighted sum of all input elements where the attention scores (α 's) represent the weighting coefficients. This allows the decoder to access all input sequence elements (the context) at each step. The key idea is that the attention weights (and context) may differ and change dynamically at each step.

The motivation behind this complicated encoder-decoder design is that we cannot translate sentences word by word. This would result in grammatically incorrect outputs, as illustrated by the RNN architecture (a) in Figure 16.2.

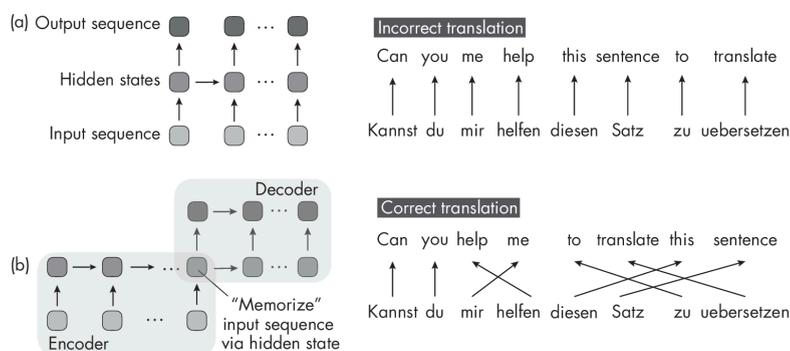


Figure 16.2

Figure 16.2 shows two different sequence-to-sequence RNN designs for sentence translation.

Figure 16.2(a) represents a regular sequence-to-sequence RNN that may be used to translate a sentence from German to English word by word.

Figure 16.2(b) depicts an encoder-decoder RNN that first reads the whole sentence before translating it.

RNN **architecture (a)** is best suited for time series tasks in which we want to make one prediction at a time, such as predicting a given stock price day by day. For tasks like language translation, we typically opt for an encoder-decoder RNN, as in **architecture (b)** in Figure 16.2. Here, the RNN encodes the input sentence, stores it in an intermediate hidden representation, and generates the output sentence. However, this creates a bottleneck where the RNN has to



memorize the whole input sentence via a single hidden state, which does not work well for longer sequences.

The bottleneck depicted in [architecture \(b\)](#) prompted the [Bahdanau](#) attention mechanism's original design, allowing the decoder to access all elements in the input sentence at each time step. The attention scores also give different weights to the different input elements depending on the current word that the decoder generates. For example, when generating the word *help* in the output sequence, the word *helfen* in the German input sentence may get a large attention weight, as it's highly relevant in this context.

The Self-Attention Mechanism

The Bahdanau attention mechanism relies on a somewhat complicated encoder-decoder design to model long-term dependencies in sequence-to-sequence language modeling tasks. Approximately three years after the Bahdanau mechanism, researchers worked on simplifying sequence-to-sequence modeling architectures by asking whether the RNN backbone was even needed to achieve good language translation performance. This led to the design of the original transformer architecture and self-attention mechanism.

Tips: transformer 架构，在 2017 年提出，用于解决序列到序列（sequence-to-sequence）语言建模任务中的长程依赖问题。

在 transformer 架构中，注意力机制被应用于同一序列中的所有元素，而不是像 Bahdanau 注意力机制那样涉及两个不同的序列。

与 RNN 中的注意力机制类似，上下文向量是输入序列元素的加权和，其中注意力分数 (α) 表示权重系数。

这允许解码器访问输入序列的所有元素（上下文）。

In self-attention, the attention mechanism is applied between all elements in the same sequence (as opposed to involving two sequences), as depicted in the simplified attention mechanism in [Figure 16.3](#). Similar to the attention mechanism for RNNs, the context vector is an attention-weighted sum over the input sequence elements.

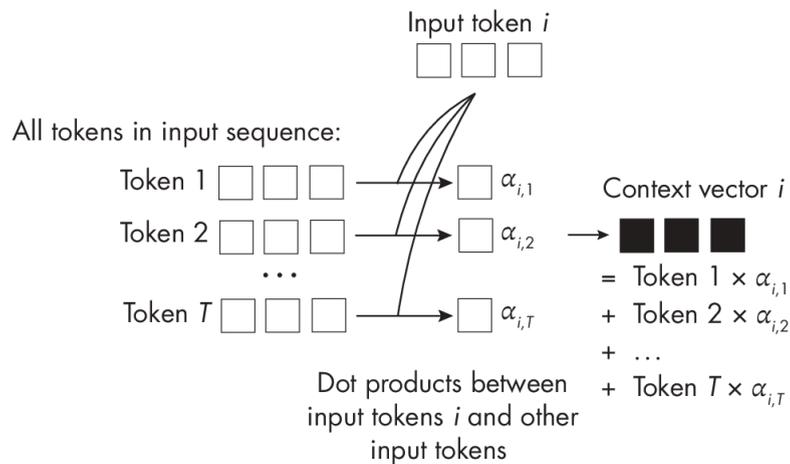


Figure 16.3

While Figure 16.3 doesn't include weight matrices, the self-attention mechanism used in transformers typically involves multiple weight matrices to compute the attention weights.

This chapter laid the groundwork for understanding the inner workings of transformer models and the attention mechanism. The next chapter covers the different types of transformer architectures in more detail.

Exercises

16-1. Considering that self-attention compares each sequence element with itself, what is the time and memory complexity of self-attention?

16-2. We discussed self-attention in the context of natural language processing. Could this mechanism be useful for computer vision applications as well?

References

- The paper introducing the original self-attention mechanism, also known as *scaled dot-product* attention: Ashish Vaswani et al., "Attention Is All You Need"? (2017), <https://arxiv.org/abs/1706.03762>.
- The Bahdanau attention mechanism for RNNs: Dzmitry Bahdanau, Kyunghyun Cho, and Yoshua Bengio, "Neural Machine Translation by Jointly Learning to Align and Translate"? (2014), <https://arxiv.org/abs/1409.0473>.

- For more about the parameterized self-attention mechanism, check out my blog post: "Understanding and Coding the Self-Attention Mechanism of Large Language Models from Scratch"? at <https://sebastianraschka.com/blog/2023/self-attention-from-scratch.html>.
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Chapter 17: Encoder- and Decoder-Style Transformers

What are the differences between encoder- and decoder-based language transformers?

Both encoder- and decoder-style architectures use the same self-attention layers to encode word tokens. The main difference is that encoders are designed to learn embeddings that can be used for various predictive modeling tasks such as classification. In contrast, decoders are designed to generate new texts, for example, to answer user queries.

Tips: 编码器（encoder）和解码器（decoder）架构都采用自注意力（self-attention）层来对词元进行编码。

主要区别在于：

- 编码器，侧重于**学习文本**的上下文表示（embedding），以便用于分类等各种预测任务；
- 而解码器，则专注于**生成**新的文本内容，比如用于回答用户问题等生成式任务。

This chapter starts by describing the original transformer architecture consisting of an encoder that processes input text and a decoder that produces translations. The subsequent sections then describe how models like BERT and RoBERTa utilize only the encoder to understand context and how the GPT architectures emphasize decoder-only mechanisms for text generation.

The Original Transformer

The original transformer architecture introduced in Chapter [\[ch16\]](#) was developed for English-to-French and English-to-German language translation. It utilized both an encoder and a decoder, as illustrated in Figure [17.1](#).

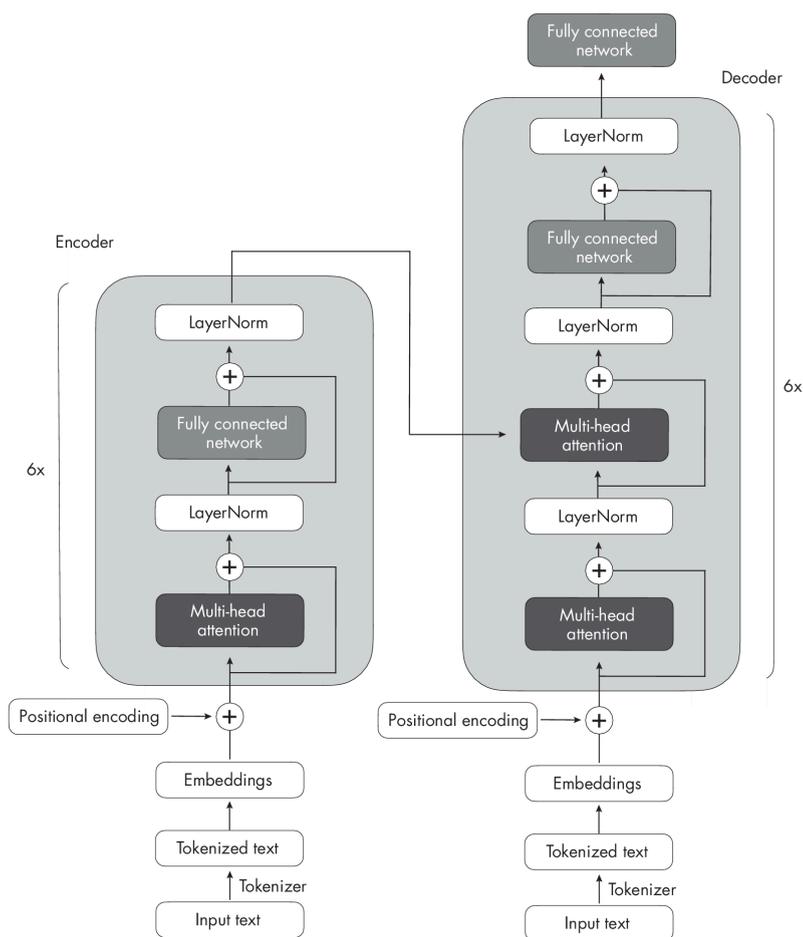


Figure 17.1

In Figure 17.1, the input text (that is, the sentences of the text to be translated) is first **tokenized** into individual word tokens, which are then encoded via an **embedding layer** before they enter the encoder part (see Chapter [ch01] for more on embeddings). After a **positional encoding vector** is added to each embedded word, the embeddings go through a **multi-head self-attention layer**. This layer is followed by an **addition step**, indicated by a plus sign (+) in Figure 17.1, which performs a **layer normalization** and adds the original embeddings via a **skip connection**, also known as a **residual** or **shortcut connection**. Following this is a LayerNorm block, short for *layernormalization*, which normalizes the activations of the previous layer to improve the stability of the neural network's training. The addition of the original embeddings and the layer normalization steps are often summarized as the **Add & Norm step**.

Finally, after entering the **fully connected network** -- a small, multilayer perceptron consisting of two fully connected layers with a nonlinear activation function in between -- the outputs are again added and normalized before they are passed to a **multi-head self-attention layer** of the decoder.



Tips:

- **tokenized** : 将文本拆分成单词或词元 (token)。
- **embedding layer** : 将词元转换为向量表示。
- **positional encoding vector** : 将词元的位置信息添加到向量表示中。
- **multi-head self-attention layer** : 对词元进行自注意力计算。
- **addition step** : 将原始向量表示和自注意力计算结果相加, 也称为**残差连接、跳跃连接、快捷连接**。
- **layer normalization** : 对每个层的激活进行归一化, 以提高神经网络的稳定性。
- **Add & Normstep** : 将原始向量表示和自注意力计算结果相加, 并进行归一化。
- **fully connected network** : 一个小的多层感知器, 包含两个全连接层和一个非线性激活函数。

The decoder in Figure [17.1](#) has a similar overall structure to the encoder. The key difference is that the inputs and outputs are different: the encoder receives the input text to be translated, while the decoder generates the translated text.

Encoders

The encoder part in the original transformer, as illustrated in Figure [17.1](#), is responsible for understanding and extracting the relevant information from the input text. It then outputs a continuous representation (embedding) of the input text, which is passed to the decoder. Finally, the decoder generates the translated text (target language) based on the continuous representation received from the encoder.

Over the years, various **encoder-only** architectures have been developed based on the encoder module of the original transformer model outlined earlier. One notable example is **BERT**, which stands for **Bidirectional Encoder Representations from Transformers**.

As noted in Chapter [\[ch14\]](#), BERT is an encoder-only architecture based on the transformer's encoder module. The BERT model is pretrained on a large text corpus using masked language modeling and next-sentence prediction tasks.



Figure 17.2 illustrates the masked language modeling pretraining objective used in BERT-style transformers.

Tips: BERT 是基于掩码语言建模和预测下一句任务预训练的模型。

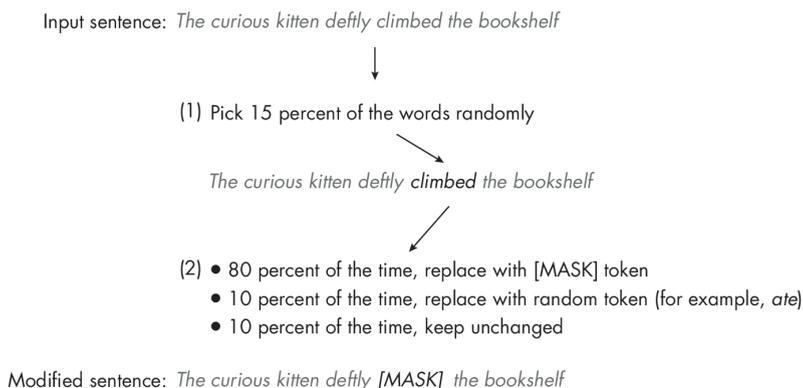


Figure 17.2

As Figure 17.2 demonstrates, the main idea behind masked language modeling is to mask (or replace) random word tokens in the input sequence and then train the model to predict the original masked tokens based on the surrounding context.

In addition to the **masked language modeling** pretraining task illustrated in Figure 17.2, the next-sentence prediction task asks the model to predict whether the original document's sentence order of two randomly shuffled sentences is correct. For example, say that two sentences, in random order, are separated by the [SEP] token (*SEP* is short for *separate*). The brackets are a part of the token's notation and are used to make it clear that this is a special token as opposed to a regular word in the text. BERT-style transformers also use a [CLS] token. The [CLS] token serves as a placeholder token for the model, prompting the model to return a *True* or *False* label indicating whether the sentences are in the correct order:

- "[CLS] Toast is a simple yet delicious food. [SEP] It's often served with butter, jam, or honey."?
- "[CLS] It's often served with butter, jam, or honey. [SEP] Toast is a simple yet delicious food."?

The masked language and next-sentence pretraining objectives allow BERT to learn rich contextual representations of the input texts, which can then be fine-tuned for various downstream tasks like sentiment analysis, question answering,



and named entity recognition. It's worth noting that this pretraining is a form of self-supervised learning (see Chapter [\[ch02\]](#) for more details on this type of learning).

Tips: 值得一提的是，这种预训练方式是一种自监督学习（self-supervised learning）

RoBERTa , which stands for **Robustly Optimized BERT Approach**, is an improved version of BERT. It maintains the same overall architecture as BERT but employs several training and optimization improvements, such as larger batch sizes, more training data, and eliminating the next-sentence prediction task. These changes have resulted in RoBERTa achieving better performance on various natural language understanding tasks than BERT.

Decoders

Coming back to the original transformer architecture outlined in Figure [17.1](#), the multi-head self-attention mechanism in the decoder is similar to the one in the encoder, but it is masked to prevent the model from attending to future positions, ensuring that the predictions for position i can depend only on the known outputs at positions less than i . As illustrated in Figure [17.3](#), the decoder generates the output word by word.

Tips: 解码器中，为了防止模型关注未来的位置上信息，会进行掩码处理。

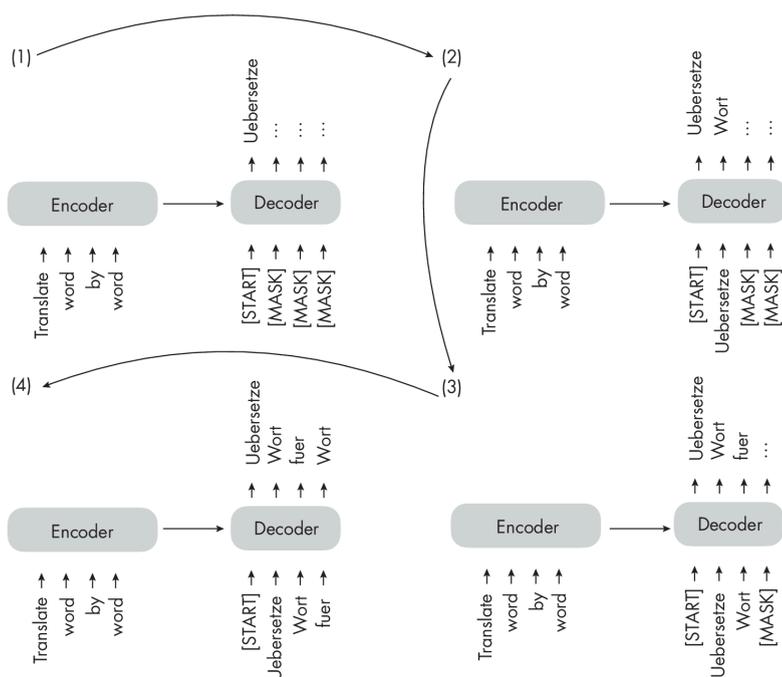


Figure 17.3

This masking (shown explicitly in Figure 17.3, although it occurs internally in the decoder's multi-head self-attention mechanism) is essential to maintaining the transformer model's autoregressive property during training and inference. This autoregressive property ensures that the model generates output tokens one at a time and uses previously generated tokens as context for generating the next word token.

Tips: 解码器中，为了 **保持模型自回归** 的特性，会进行 **掩码** 处理。

自回归 (Autoregressive): 在生成式任务中，模型会根据之前生成的内容，预测下一个词。

FIXME: 自回归 auto-regressive 是什么意思???

Over the years, researchers have built upon the original encoder-decoder transformer architecture and developed several decoder-only models that have proven highly effective in various natural language processing tasks. The most notable models include the GPT family, which we briefly discussed in Chapter [ch14] and in various other chapters throughout the book. **GPT** stands for **Generative Pre-trained Transformer**. The GPT series comprises decoder-only models pretrained on large-scale unsupervised text data and fine-tuned for specific tasks such as text classification, sentiment analysis, question answering, and summarization. The GPT models, including at the time of writing GPT-2, GPT-



3, and GPT-4, have shown remarkable performance in various benchmarks and are currently the most popular architecture for natural language processing.

One of the most notable aspects of GPT models is their **emergent** properties. **Emergent** properties are the abilities and skills that a model develops due to its next-word prediction pretraining. Even though these models were taught only to predict the next word, the pretrained models are capable of text summarization, translation, question answering, classification, and more. Furthermore, these models can perform new tasks without updating the model parameters via in-context learning, which we'll discuss in more detail in Chapter [\[ch18\]](#).

Encoder-Decoder Hybrids

Next to the traditional encoder and decoder architectures, there have been advancements in the development of new encoder-decoder models that leverage the strengths of both components. These models often incorporate novel techniques, pretraining objectives, or architectural modifications to enhance their performance in various natural language processing tasks. Some notable examples of these new encoder-decoder models include **BART** and **T5**.

Encoder-decoder models are typically used for natural language processing tasks that involve understanding input sequences and generating output sequences, often with different lengths and structures. They are particularly good at tasks where there is a complex mapping between the input and output sequences and where it is crucial to capture the relationships between the elements in both sequences. Some common use cases for encoder-decoder models include text translation and summarization.

Tips: Encoder-decoder models, 通常用于自然语言处理任务，涉及理解输入序列和生成输出序列，通常具有不同的长度和结构。在文本翻译和摘要等任务中，表现尤为出色。

Terminology

All of these methods -- **encoder-only**, **decoder-only**, and **encoder-decoder** models -- are **sequence-to-sequence** models (often abbreviated as



`seq2seq`). While we refer to BERT-style methods as "encoder-only," the description may be misleading since these methods also *decode* the embeddings into output tokens or text during pretraining. In other words, both encoder-only and decoder-only architectures perform decoding.

However, the encoder-only architectures, in contrast to decoder-only and encoder-decoder architectures, don't decode in an autoregressive fashion. *Autoregressive decoding* refers to generating output sequences one token at a time, conditioning each token on the previously generated tokens. Encoder-only models do not generate coherent output sequences in this manner. Instead, they focus on understanding the input text and producing task-specific outputs, such as labels or token predictions.

Contemporary Transformer Models

In brief, encoder-style models are popular for learning embeddings used in classification tasks, encoder-decoder models are used in generative tasks where the output heavily relies on the input (for example, translation and summarization), and decoder-only models are used for other types of generative tasks, including Q&A. Since the first transformer architecture emerged, hundreds of encoder-only, decoder-only, and encoder-decoder hybrids have been developed, as diagrammed in Figure [17.4](#).

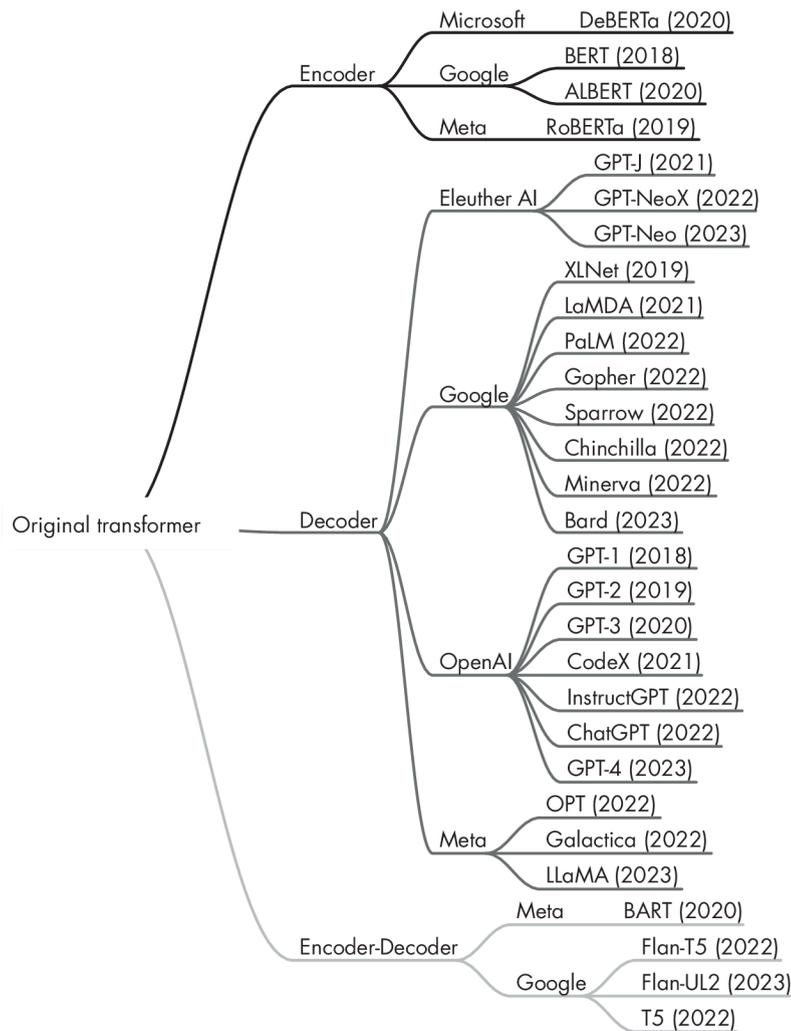


Figure 17.4

While encoder-only models have gradually become less popular, decoder-only models like GPT have exploded in popularity, thanks to breakthroughs in text generation via GPT-3, ChatGPT, and GPT-4. However, encoder-only models are still very useful for training predictive models based on text embeddings as opposed to generating texts.

Exercises

17-1. As discussed in this chapter, BERT-style encoder models are pretrained using masked language modeling and next-sentence prediction pretraining objectives. How could we adopt such a pretrained model for a classification task (for example, predicting whether a text has a positive or negative sentiment)?

17-2. Can we fine-tune a decoder-only model like GPT for classification?



References

- The Bahdanau attention mechanism for RNNs: Dzmitry Bahdanau, Kyunghyun Cho, and Yoshua Bengio, "Neural Machine Translation by Jointly Learning to Align and Translate"? (2014), <https://arxiv.org/abs/1409.0473>.
- The original BERT paper, which popularized encoder-style transformers with a masked word and a next-sentence prediction pretraining objective: Jacob Devlin et al., "BERT: Pre-training of Deep Bidirectional Transformers for Language Understanding"? (2018), <https://arxiv.org/abs/1810.04805>.
- RoBERTa improves upon BERT by optimizing training procedures, using larger training datasets, and sentence prediction task: Yinhan Liu et al., "RoBERTa: A Robustly Optimized BERT Pretraining Approach"? (2019), <https://arxiv.org/abs/1907.11692>.
- The BART encoder-decoder architecture: Mike Lewis et al., "BART: Denoising Sequence-to-Sequence Pre-training for Natural Language Generation, Translation, and Comprehension"? (2018), <https://arxiv.org/abs/1910.13461>.
- The T5 encoder-decoder architecture: Colin Raffel et al., "Exploring the Limits of Transfer Learning with a Unified Text-to-Text Transformer"? (2019), <https://arxiv.org/abs/1910.10683>.
- The paper proposing the first GPT architecture: Alec Radford et al., "Improving Language Understanding by Generative Pre-Training"? (2018), https://cdn.openai.com/research-covers/language-unsupervised/language_understanding_paper.pdf.
- The GPT-2 model: Alec Radford et al., "Language Models Are Unsupervised Multitask Learners"? (2019), <https://www.semanticscholar.org/paper/Language-Models-are-Unsupervised-Multitask-Learners-Radford-Wu/9405cc0d6169988371b2755e573cc28650d14dfe>.
- The GPT-3 model: Tom B. Brown et al., "Language Models Are Few-Shot Learners"? (2020), <https://arxiv.org/abs/2005.14165>.



Chapter 18: Using and Fine-Tuning Pretrained Transformers

What are the different ways to use and fine-tune pretrained large language models?

使用和微调预训练大语言模型的三种方法：

1. 特征提取
2. 上下文学习（提示学习）
3. 微调部分模型参数

The three most common ways to use and fine-tune pretrained LLMs include a **feature-based approach** , **in-context prompting** , and **updating a subset of the model parameters** . First, most pretrained LLMs or language transformers can be utilized without the need for further fine-tuning. For instance, we can employ a feature-based method to train a new downstream model, such as a linear classifier, using embeddings generated by a pretrained transformer. Second, we can showcase examples of a new task within the input itself, which means we can directly exhibit the expected outcomes without requiring any updates or learning from the model. This concept is also known as *prompting*. Finally, it's also possible to fine-tune all or just a small number of parameters to achieve the desired outcomes.

The following sections discuss these types of approaches in greater depth.

Using Transformers for Classification Tasks

Let's start with the conventional methods for utilizing pretrained transformers: training another model on feature embeddings, fine-tuning output layers, and fine-tuning all layers. We'll discuss these in the context of classification. (We will revisit prompting later in the section "In-Context Learning, Indexing, and Prompt Tuning" on page .)

In the feature-based approach, we load the pretrained model and keep it "frozen," meaning we do not update any parameters of the pretrained model. Instead, we



treat the model as a feature extractor that we apply to our new dataset. We then train a downstream model on these embeddings.

特征提取方法：

1. 加载预训练模型并保持冻结状态，不更新任何参数。
2. 将预训练模型视为特征提取器，应用于新数据集。
3. 在提取的特征上训练下游模型。

This downstream model can be any model like (random forests, XGBoost, and so on), but linear classifiers typically perform best. This is likely because pretrained transformers like BERT and GPT already extract high-quality, informative features from the input data. These feature embeddings often capture complex relationships and patterns, making it easy for a linear classifier to effectively separate the data into different classes. Furthermore, linear classifiers, such as logistic regression machines and support vector machines, tend to have strong regularization properties. These regularization properties help prevent overfitting when working with high-dimensional feature spaces generated by pretrained transformers. This feature-based approach is the most efficient method since it doesn't require updating the transformer model at all. Finally, the embeddings can be precomputed for a given training dataset (since they don't change) when training a classifier for multiple training epochs.

特征提取方法的优点：

1. 不需要更新预训练模型。
2. 提取的特征通常能够捕捉复杂的关系和模式，使得线性分类器能够有效地分离数据。
3. 线性分类器通常具有较强的正则化属性，有助于防止在高维特征空间中过拟合。
4. 特征可以预先计算，用于多个训练轮次的分类器训练。

Figure [18.1](#) illustrates how LLMs are typically created and adopted for downstream tasks using fine-tuning. Here, a pretrained model, trained on a general text corpus, is fine-tuned to perform tasks like German-to-English translation.

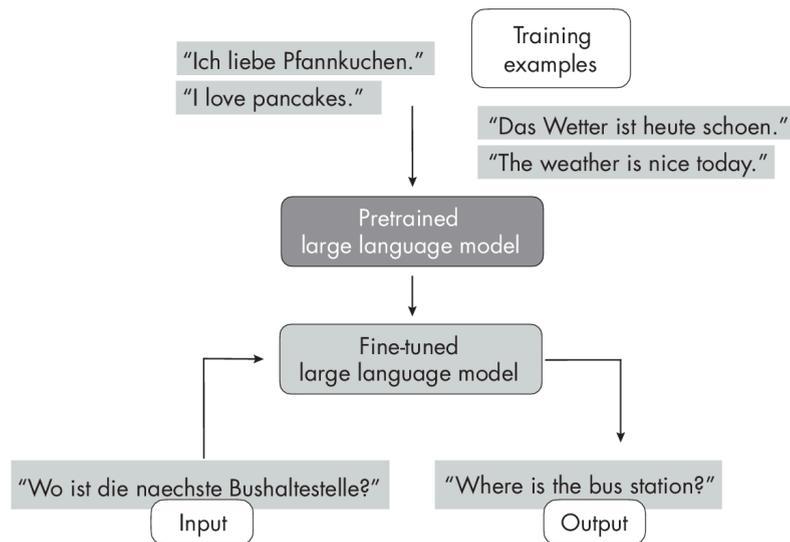


Figure 18.1

The conventional methods for fine-tuning pretrained LLMs include updating only the output layers, a method we'll refer to as *fine-tuning I*, and updating all layers, which we'll call *fine-tuning II*.

微调方法，分为 2 类：

1. 只更新输出层，称为 **fine-tuning I**。
2. 更新所有层，称为 **fine-tuning II**。

Fine-tuning I is similar to the feature-based approach described earlier, but it adds one or more output layers to the LLM itself. The backbone of the LLM remains frozen, and we update only the model parameters in these new layers. Since we don't need to backpropagate through the whole network, this approach is relatively efficient regarding throughput and memory requirements.

In **fine-tuning II**, we load the model and add one or more output layers, similarly to fine-tuning I. However, instead of backpropagating only through the last layers, we update *all* layers via backpropagation, making this the most expensive approach. While this method is computationally more expensive than the feature-based approach and fine-tuning I, it typically leads to better modeling or predictive performance. This is especially true for more specialized domain-specific datasets.

Figure 18.2 summarizes the three approaches described in this section so far.

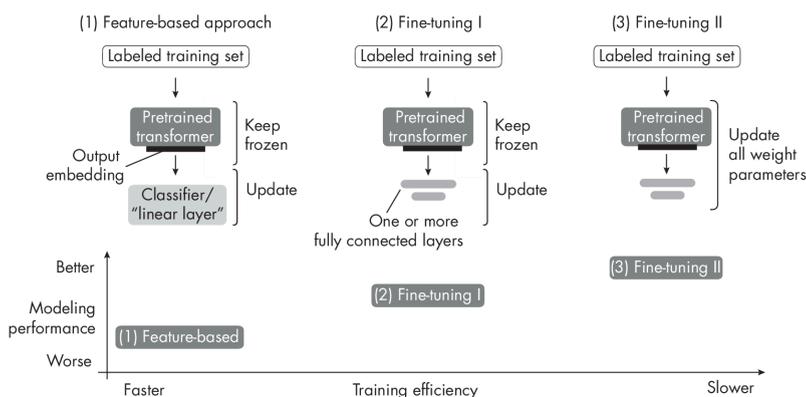


Figure 18.2

In addition to the conceptual summary of the three fine-tuning methods described in this section, Figure 18.2 also provides a rule-of-thumb guideline for these methods regarding training efficiency. Since fine-tuning II involves updating more layers and parameters than fine-tuning I, backpropagation is costlier for fine-tuning II. For similar reasons, fine-tuning II is costlier than a simpler feature-based approach.

In-Context Learning, Indexing, and Prompt Tuning

LLMs like GPT-2 and GPT-3 popularized the concept of **in-context learning**, often called **zero-shot** or **few-shot learning** in this context, which is illustrated in Figure 18.3.

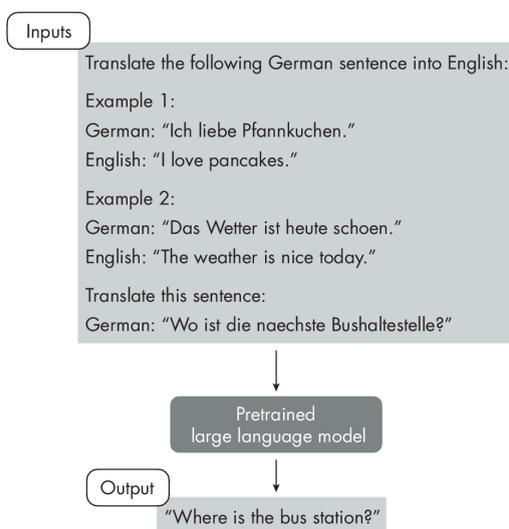


Figure 18.3



As Figure [18.3](#) shows, **in-context learning** aims to provide context or examples of the task within the input or prompt, allowing the model to infer the desired behavior and generate appropriate responses. This approach takes advantage of the model's ability to learn from vast amounts of data during pretraining, which includes diverse tasks and contexts.

The definition of **few-shot learning**, considered synonymous with in-context learning-based methods, differs from the conventional approach to few-shot learning discussed in Chapter [\[ch03\]](#).

此处讨论的 **few-shot learning** 与第 3 章讨论的 **few-shot learning** 不同。

For example, suppose we want to use in-context learning for few-shot German -- English translation using a large-scale pretrained language model like GPT-3. To do so, we provide a few examples of German -- English translations to help the model understand the desired task, as follows:

```
text

Translate the following German sentences into English:

Example 1:
German: "Ich liebe Pfannkuchen."
English: "I love pancakes."

Example 2:
German: "Das Wetter ist heute schoen."
English: "The weather is nice today."

Translate this sentence:
German: "Wo ist die naechste Bushaltestelle?"
```

Generally, in-context learning does not perform as well as fine-tuning for certain tasks or specific datasets since it relies on the pretrained model's ability to generalize from its training data without further adapting its parameters for the particular task at hand.

上下文学习在某些任务或特定数据集上可能不如微调，因为它依赖于预训练模型从其训练数据中泛化，而无需为特定任务进一步调整其参数。



However, in-context learning has its advantages. It can be particularly useful when labeled data for fine-tuning is limited or unavailable. It also enables rapid experimentation with different tasks without fine-tuning the model parameters in cases where we don't have direct access to the model or where we interact only with the model through a UI or API (for example, ChatGPT).

上下文学习的优点：

1. 在有标签数据有限或不可用的情况下，上下文学习特别有用。
2. 在不需要直接访问模型或仅通过 UI 或 API 与模型交互的情况下，可以快速尝试不同的任务。

Related to in-context learning is the concept of **hard prompt tuning**, where *hard* refers to the non-differentiable nature of the input tokens. Where the previously described fine-tuning methods update the model parameters to better perform the task at hand, **hard prompt tuning** aims to optimize the prompt itself to achieve better performance. Prompt tuning does not modify the model parameters, but it may involve using a smaller labeled dataset to identify the best prompt formulation for the specific task.

提示词工程，即提示词微调。不改变模型参数，而是优化提示词（可能包含一小部分标签示例数据），以达到更好的性能。

For example, to improve the prompts for the previous German -- English translation task, we might try the following three prompting variations:

- Translate the German sentence '{german_sentence}' into English: {english_translation}
- German: '{german_sentence}' English: {english_translation}
- From German to English: '{german_sentence}' -> {english_translation}

提示词工程的优点：

1. 资源效率高，不需要更新模型参数。
2. 性能通常不如全模型微调，因为它不更新模型参数，可能限制其适应特定任务的细微差别。



Prompt tuning is a resource-efficient alternative to parameter fine-tuning. However, its performance is usually not as good as full model fine-tuning, as it does not update the model's parameters for a specific task, potentially limiting its ability to adapt to task-specific nuances. Furthermore, prompt tuning can be labor intensive since it requires either human involvement comparing the quality of the different prompts or another similar method to do so. This is often known as *hard* prompting since, again, the input tokens are not differentiable. In addition, other methods exist that propose to use another LLM for automatic prompt generation and evaluation.

Yet another way to leverage a purely in-context learning-based approach is

indexing, illustrated in Figure 18.4

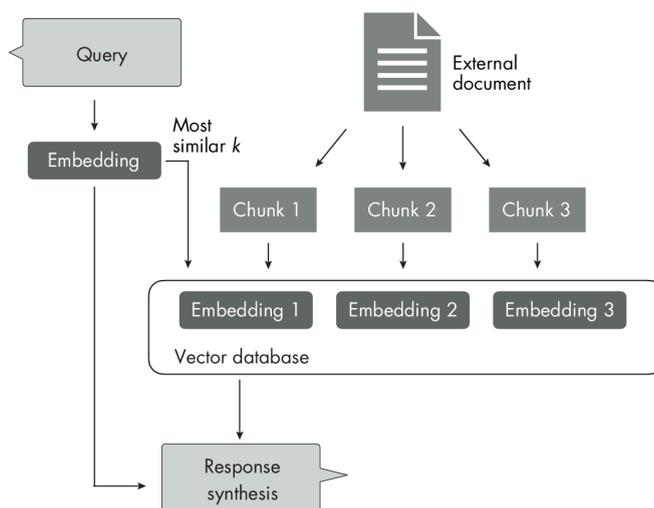


Figure 18.4

In the context of LLMs, we can think of **indexing** as a workaround based on in-context learning that allows us to turn LLMs into information retrieval systems to extract information from external resources and websites. In Figure 18.4, an indexing module parses a document or website into smaller chunks, embedded into vectors that can be stored in a vector database. When a user submits a query, the indexing module computes the vector similarity between the embedded query and each vector stored in the database. Finally, the indexing module retrieves the top k most similar embeddings to synthesize the response.

索引，即索引模块，将文档或网站解析为更小的块 chunk，嵌入到向量中，可以存储在向量数据库中。

- 当用户提交查询时，索引模块计算嵌入查询与数据库中每个向量的相似度。
- 最后，索引模块检索与查询最相似的 k 个嵌入向量，以合成响应。



Parameter-Efficient Fine-Tuning

In recent years, many methods have been developed to adapt pretrained transformers more efficiently for new target tasks. These methods are commonly referred to as **parameter-efficient fine-tuning**, with the most popular methods at the time of writing summarized in Figure 18.5.

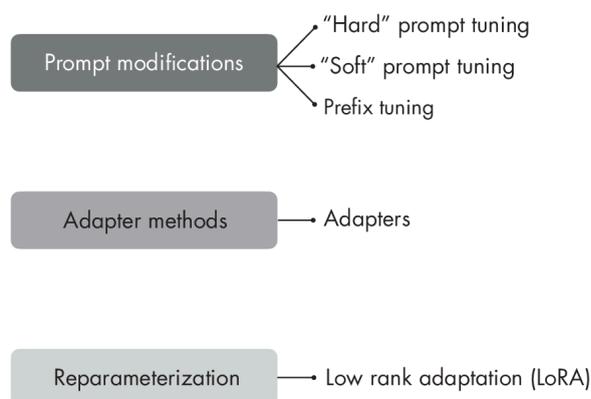


Figure 18.5

In contrast to the hard prompting approach discussed in the previous section, **soft prompting** strategies optimize embedded versions of the prompts. While in **hard prompt** tuning we modify the discrete input tokens, in **soft prompt** tuning we utilize trainable parameter tensors instead. The idea behind soft prompt tuning is to prepend a trainable parameter tensor (the "soft prompt") to the embedded query tokens. The prepended tensor is then tuned to improve the modeling performance on a target dataset using gradient descent.

硬提示词，调整了输入的离散 tokens；软提示词，调整了输入的 tokens 的嵌入。

- 软提示词的思路是，在输入的 tokens 前添加一个可训练的参数 tensor（即软提示词），
- 然后使用梯度下降优化这个 tensor，以提高在目标数据集上的建模性能。



In Python-like pseudocode, soft prompt tuning can be described as

```
python
x = EmbeddingLayer(input_ids)
x = concatenate([soft_prompt_tensor, x], dim=seq_len)
output = model(x)
```

where the `soft_prompt_tensor` has the same feature dimension as the embedded inputs produced by the embedding layer. Consequently, the modified input matrix has additional rows (as if it extended the original input sequence with additional tokens, making it longer).

Another popular prompt tuning method is `prefix tuning`. *Prefix tuning* is similar to soft prompt tuning, except that in prefix tuning, we prepend trainable tensors (soft prompts) to each transformer block instead of only the embedded inputs, which can stabilize the training.

前缀调优，在每个 transformer block 前添加一个可训练的参数 tensor。

The implementation of prefix tuning is illustrated in the following pseudocode:

```
python
def transformer_block_with_prefix(x):
    soft_prompt = FullyConnectedLayers(# Prefix
        soft_prompt)                  # Prefix
    x = concatenate([soft_prompt, x], # Prefix
                    dim=seq_len)     # Prefix

    residual = x
    x = SelfAttention(x)
    x = LayerNorm(x + residual)
    residual = x
    x = FullyConnectedLayers(x)
    x = LayerNorm(x + residual)
    return x
```

Let's break Listing [18.6](#) into three main parts: implementing the soft prompt, concatenating the soft prompt (prefix) with the input, and implementing the rest



of the transformer block.

First, the `soft_prompt`, a tensor, is processed through a set of fully connected layers. Second, the transformed soft prompt is concatenated with the main input, `x`. The dimension along which they are concatenated is denoted by `seq_len`, referring to the sequence length dimension. Third, the subsequent lines of code describe the standard operations in a transformer block, including self-attention, layer normalization, and feed-forward neural network layers, wrapped around residual connections.

As shown in Listing [18.6](#), `prefix tuning` modifies a transformer block by adding a trainable `soft prompt`. Figure [18.6](#) further illustrates the difference between a regular transformer block and a prefix tuning transformer block.

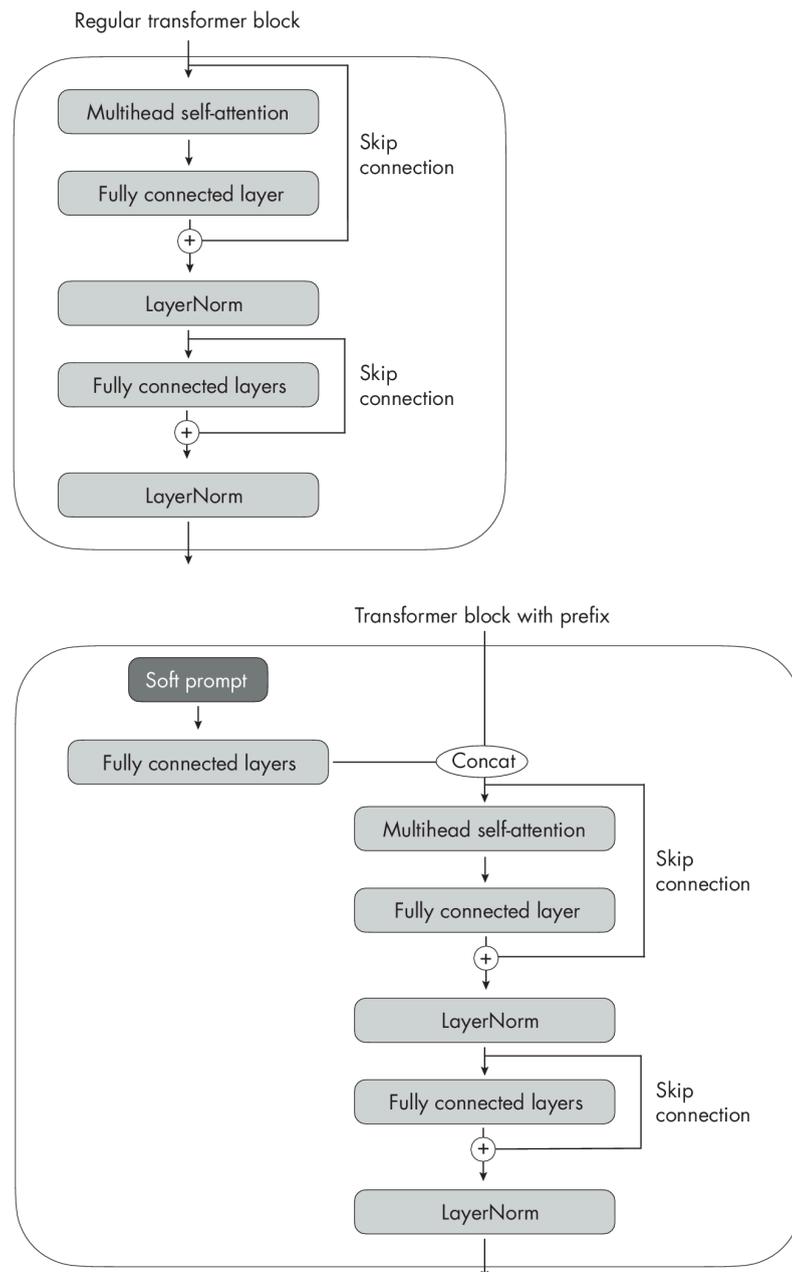


Figure 18.6

Both **soft prompt** tuning and **prefix tuning** are considered parameter efficient since they require training only the prepended parameter tensors and not the LLM parameters themselves.

Adapter methods are related to **prefix tuning** in that they add additional parameters to the transformer layers. In the original adapter method, additional fully connected layers were added after the multihead self-attention and existing fully connected layers in each transformer block, as illustrated in Figure [18.7](#).

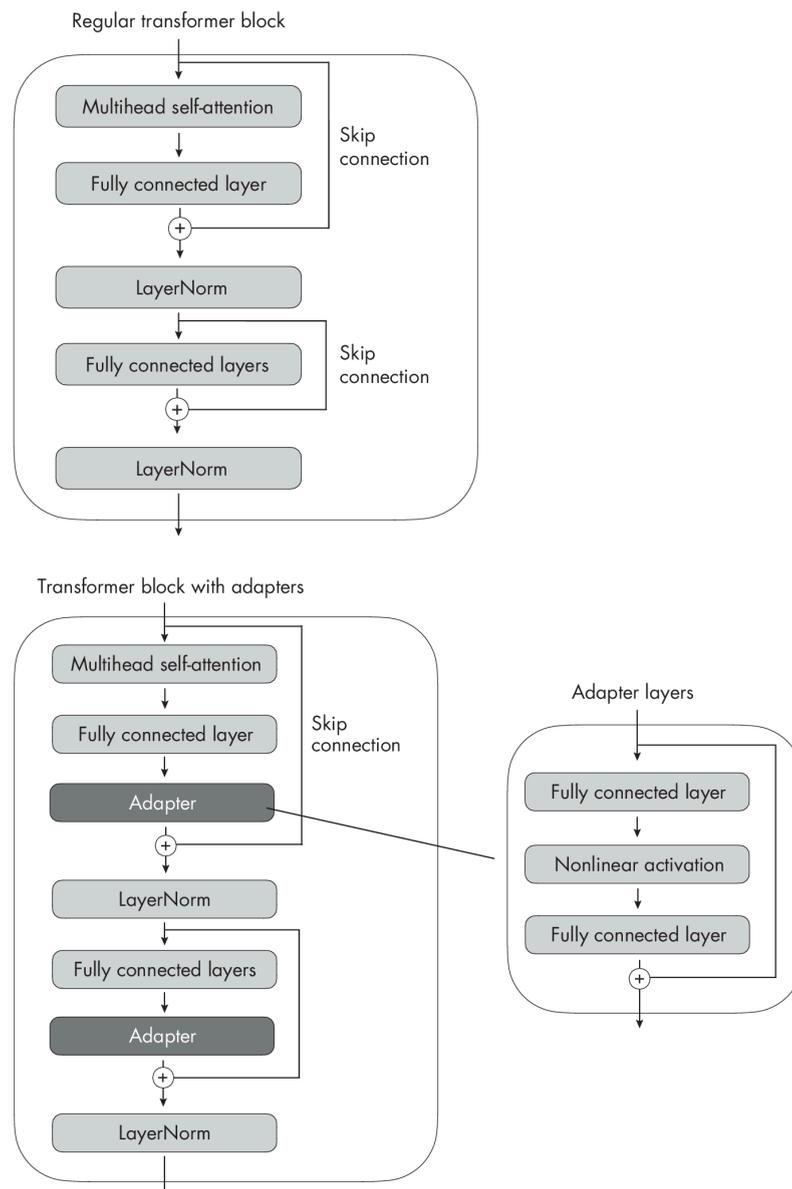


Figure 18.7

Only the new adapter layers are updated when training the LLM using the original adapter method, while the remaining transformer layers remain frozen. Since the adapter layers are usually small -- the first fully connected layer in an adapter block projects its input into a low-dimensional representation, while the second layer projects it back into the original input dimension -- this adapter method is usually considered parameter efficient.

只会更新 adapter 层，其他层保持冻结。

In pseudocode, the original adapter method can be written as follows:

python

```
def transformer_block_with_adapter(x):
    residual = x
    x = SelfAttention(x)
    x = FullyConnectedLayers(x) # Adapter
    x = LayerNorm(x + residual)
    residual = x
    x = FullyConnectedLayers(x)
    x = FullyConnectedLayers(x) # Adapter
    x = LayerNorm(x + residual)
    return x
```

Low-rank adaptation (LoRA), another popular parameter-efficient fine-tuning method worth considering, refers to reparameterizing pretrained LLM weights using low-rank transformations. LoRA is related to the concept of **low-rank transformation**, a technique to approximate a high-dimensional matrix or dataset using a lower-dimensional representation. The lower-dimensional representation (or **low-rank approximation**) is achieved by finding a combination of fewer dimensions that can effectively capture most of the information in the original data. Popular low-rank transformation techniques include **principal component analysis** and **singular vector decomposition**.

低秩适应 (LoRA), 另一种流行的参数高效微调方法, 值得考虑, 指的是使用低秩变换重新参数化预训练 LLM 权重。

- LoRA 与低秩变换的概念相关, 低秩变换是一种技术, 使用较低维度的表示来近似高维矩阵或数据集。
- 低秩变换 (或低秩近似) 通过找到更少的维度组合来有效捕获原始数据中的大部分信息。
- 流行的低秩变换技术包括主成份分析和奇异值分解。

For example, suppose ΔW represents the parameter update for a weight matrix of the LLM with dimension $\mathbb{R}^{A \times B}$. We can decompose the weight update matrix into two smaller matrices: $\Delta W = W_A W_B$, where $W_A \in \mathbb{R}^{A \times h}$ and $W_B \in \mathbb{R}^{h \times B}$. Here, we keep the original weight frozen and train only the new matrices W_A and W_B .



How is this method parameter efficient if we introduce new weight matrices?

These new matrices can be very small. For example, if $A = 25$ and $B = 50$, then the size of ΔW is $25 \times 50 = 1,250$. If $h = 5$, then W_A has 125 parameters, W_B has 250 parameters, and the two matrices combined have only $125 + 250 = 375$ parameters in total.

After learning the weight update matrix, we can then write the matrix multiplication of a fully connected layer, as shown in this pseudocode:

```
python
def lora_forward_matmul(x):
    h = x . W # Regular matrix multiplication
    h += x . (W_A . W_B) * scalar
    return h
```

In Listing [\[matrixMultiplication\]](#), `scalar` is a scaling factor that adjusts the magnitude of the combined result (original model output plus low-rank adaptation). This balances the pretrained model's knowledge and the new task-specific adaptation.

According to the original paper introducing the LoRA method, models using LoRA perform slightly better than models using the adapter method across several task-specific benchmarks. Often, LoRA performs even better than models fine-tuned using the fine-tuning II method described earlier.

原始提出 LoRA 方法的论文指出，使用 LoRA 的模型在多个任务特定基准上略微优于使用适配器方法的模型。通常，LoRA 甚至比前面描述的 fine-tuning II 方法微调的模型性能更好。

Reinforcement Learning with Human Feedback

The previous section focused on ways to make fine-tuning more efficient. Switching gears, how can we improve the modeling performance of LLMs via fine-tuning?



The conventional way to adapt or fine-tune an LLM for a new target domain or task is to use a supervised approach with labeled target data. For instance, the **fine-tuning II** approach allows us to adapt a pretrained LLM and fine-tune it on a target task such as sentiment classification, using a dataset that contains texts with sentiment labels like *positive*, *neutral*, and *negative*.

监督微调，用有标签的目标数据集，训练 LLM 以适应新任务。

Supervised fine-tuning is a foundational step in training an LLM. An additional, more advanced step is **reinforcement learning with human feedback (RLHF)**, which can be used to further improve the model's alignment with human preferences. For example, ChatGPT and its predecessor, InstructGPT, are two popular examples of pretrained LLMs (GPT-3) fine-tuned using RLHF.

强化学习，用人类反馈，训练 LLM 以适应人类偏好。

In **RLHF**, a pretrained model is fine-tuned using a combination of supervised learning and reinforcement learning. This approach was popularized by the original ChatGPT model, which was in turn based on InstructGPT. Human feedback is collected by having humans rank or rate different model outputs, providing a reward signal. The collected reward labels can be used to train a reward model that is then used to guide the LLMs' adaptation to human preferences. The reward model is learned via supervised learning, typically using a pretrained LLM as the base model, and is then used to adapt the pretrained LLM to human preferences via additional fine-tuning. The training in this additional fine-tuning stage uses a flavor of reinforcement learning called **proximal policy optimization (PPO)**.

在 RLHF 中，使用监督学习和强化学习相结合的方法，训练 LLM 以适应人类偏好。

- 人类反馈被收集，通过人类对不同模型输出的排序或评分，提供奖励信号。
- 收集到的奖励标签可以用于训练奖励模型，然后用于指导 LLM 适应人类偏好。
- 奖励模型通过监督学习学习，通常使用预训练 LLM 作为基础模型，然后用于进一步微调预训练 LLM 以适应人类偏好。

- 在额外的微调阶段，使用一种称为 **proximal policy optimization** 的强化学习方法进行训练。



RLHF uses a reward model instead of training the pretrained model on the human feedback directly because involving humans in the learning process would create a bottleneck since we cannot obtain feedback in realtime.

使用**奖励模型**而不是直接在人类反馈上训练预训练模型，因为涉及人类的学习过程会创建瓶颈，因为无法实时获得反馈。

Adapting Pretrained Language Models

While fine-tuning all layers of a pretrained LLM remains the gold standard for adaption to new target tasks, several efficient alternatives exist for leveraging pretrained transformers. For instance, we can effectively apply LLMs to new tasks while minimizing computational costs and resources by utilizing feature-based methods, in-context learning, or parameter-efficient fine-tuning techniques.

The three conventional methods -- feature-based approach, fine-tuning I, and fine-tuning II -- provide different computational efficiency and performance trade-offs. Parameter-efficient fine-tuning methods like soft prompt tuning, prefix tuning, and adapter methods further optimize the adaptation process, reducing the number of parameters to be updated. Meanwhile, RLHF presents an alternative approach to supervised fine-tuning, potentially improving modeling performance.

三种传统方法 -- 特征基方法、微调 I 和微调 II -- 提供了不同的计算效率和性能权衡。

- 参数高效微调方法，如软提示词调优、前缀调优和适配器方法，进一步优化了适配过程，减少了需要更新的参数数量。
- RLHF 提供了一种替代监督微调的方法，可能提高建模性能。

In sum, the versatility and efficiency of pretrained LLMs continue to advance, offering new opportunities and strategies for effectively adapting these models to a wide array of tasks and domains. As research in this area progresses, we can



expect further improvements and innovations in using pretrained language models.

Exercises

18-1. When does it make more sense to use in-context learning rather than fine-tuning, and vice versa?

18-2. In prefix tuning, adapters, and LoRA, how can we ensure that the model preserves (and does not forget) the original knowledge?

References

- The paper introducing the GPT-2 model: Alec Radford et al., "Language Models Are Unsupervised Multitask Learners"? (2019), [https://www.semanticscholar.org/paper/Language-Models-are-Unsupervised - Multitask-Learners-Radford-Wu/9405cc0d6169988371b2755e573](https://www.semanticscholar.org/paper/Language-Models-are-Unsupervised-Multitask-Learners-Radford-Wu/9405cc0d6169988371b2755e573).
- The paper introducing the GPT-3 model: Tom B. Brown et al., "Language Models Are Few-Shot Learners"? (2020), <https://arxiv.org/abs/2005.14165>.
- The automatic prompt engineering method, which proposes using another LLM for automatic prompt generation and evaluation: Yongchao Zhou et al., "Large Language Models Are Human-Level Prompt Engineers"? (2023), <https://arxiv.org/abs/2211.01910>.
- LlamaIndex is an example of an indexing approach that leverages in-context learning: https://github.com/jerryliu/llama_index.
- DSPy is a popular open source library for retrieval augmentation and indexing: <https://github.com/stanfordnlp/dsp>.
- A first instance of soft prompting: Brian Lester, Rami Al-Rfou, and Noah Constant, "The Power of Scale for Parameter-Efficient Prompt Tuning"? (2021), <https://arxiv.org/abs/2104.08691>.
- The paper that first described prefix tuning: Xiang Lisa Li and Percy Liang, "Prefix-Tuning: Optimizing Continuous Prompts for Generation"? (2021),



<https://arxiv.org/abs/2101.00190>.

- The paper introducing the original adapter method: Neil Houlsby et al., "Parameter-Efficient Transfer Learning for NLP"? (2019)
<https://arxiv.org/abs/1902.00751>.
 - The paper introducing the LoRA method: Edward J. Hu et al., "LoRA: Low-Rank Adaptation of Large Language Models"? (2021),
<https://arxiv.org/abs/2106.09685>.
 - A survey of more than 40 research papers covering parameter-efficient fine-tuning methods: Vladislav Lialin, Vijeta Deshpande, and Anna Rumshisky, "Scaling Down to Scale Up: A Guide to Parameter-Efficient Fine-Tuning"? (2023), <https://arxiv.org/abs/2303.15647>.
 - The InstructGPT paper: Long Ouyang et al., "Training Language Models to Follow Instructions with Human Feedback"? (2022),
<https://arxiv.org/abs/2203.02155>.
 - Proximal policy optimization, which is used for reinforcement learning with human feedback: John Schulman et al., "Proximal Policy Optimization Algorithms"? (2017), <https://arxiv.org/abs/1707.06347>.
-



Chapter 19: Evaluating Generative Large Language Models

What are the standard metrics for evaluating the quality of text generated by large language models, and why are these metrics useful?

Perplexity , **BLEU** , **ROUGE** , and **BERTScore** are some of the most common evaluation metrics used in natural language processing to assess the performance of LLMs across various tasks. Although there is ultimately no way around human quality judgments, human evaluations are tedious, expensive, hard to automate, and subjective. Hence, we develop metrics to provide objective summary scores to measure progress and compare different approaches.

最终，还是需要人工评估，但是，人工评估，费时费力，且主观性较强。因此，需要开发一些客观的评估指标，来衡量模型的性能。

This chapter discusses the difference between intrinsic and extrinsic performance metrics for evaluating LLMs, and then it dives deeper into popular metrics like BLEU, ROUGE, and BERTScore and provides simple hands-on examples for illustration purposes.

本章，主要介绍，如何使用 **Perplexity** , **BLEU** , **ROUGE** , and **BERTScore** 来评估 LLM 的性能。

Evaluation Metrics for LLMs

The **perplexity metric** is directly related to the loss function used for pretraining LLMs and is commonly used to evaluate text generation and text completion models. Essentially, it quantifies the average uncertainty of the model in predicting the next word in a given context -- the lower the perplexity, the better.

困惑度，是衡量模型在给定上下文下，预测下一个词的平均不确定性。困惑度越低，模型越好。



The **bilingual evaluation understudy (BLEU)** score is a widely used metric for evaluating the quality of machine-generated translations. It measures the overlap of n-grams between the machine-generated translation and a set of human-generated reference translations. A higher BLEU score indicates better performance, ranging from 0 (worst) to 1 (best).

BLEU 是评估机器翻译质量的常用指标，它衡量机器翻译与一组人工翻译的 n-gram 重叠程度。BLEU 得分越高，表示翻译质量越好，得分范围从 0（最差）到 1（最佳）。

The **recall-oriented understudy for gisting evaluation (ROUGE)** score is a metric primarily used for evaluating automatic summarization (and sometimes machine translation) models. It measures the overlap between the generated summary and reference summaries.

ROUGE 是评估自动摘要质量的常用指标，它衡量生成摘要与一组人工摘要的重叠程度。

We can think of perplexity as an **intrinsic metric** and BLEU and ROUGE as **extrinsic metrics**. To illustrate the difference between the two types of metrics, think of optimizing the conventional cross entropy to train an image classifier. The cross entropy is a loss function we optimize during training, but our end goal is to maximize the classification accuracy. Since classification accuracy cannot be optimized directly during training, as it's not differentiable, we minimize the surrogate loss function like the cross entropy. Minimizing the cross entropy loss more or less correlates with maximizing the classification accuracy.

我们可以将困惑度视为一个 **内生指标**，而 BLEU 和 ROUGE 视为 **外生指标**。

- 为了说明这两种指标之间的差异，可以考虑优化传统的交叉熵来训练图像分类器。
- 交叉熵是一个在训练过程中优化的损失函数，但我们的最终目标是最大化分类准确率。
- 由于分类准确率不能在训练过程中直接优化，因为它不是可微的，因此我们最小化替代损失函数，如交叉熵。
- 最小化交叉熵损失，与最大化分类准确率相关。



Perplexity is often used as an evaluation metric to compare the performance of different language models, but it is not the optimization target during training. BLEU and ROUGE are more related to classification accuracy, or rather precision and recall. In fact, BLEU is a precision-like score to evaluate the quality of a translated text, while ROUGE is a recall-like score to evaluate summarized texts.

困惑度，通常用于评估不同语言模型的性能，但不是训练的目标。

- BLEU 和 ROUGE 与分类准确率相关，或者更准确地说，与精确率和召回率相关。
- 事实上，BLEU 是一个类似于精确率的评分，用于评估翻译文本的质量，而 ROUGE 是一个类似于召回率的评分，用于评估摘要文本的质量。

The following sections discuss the mechanics of these metrics in more detail.

Perplexity

Perplexity is closely related to the **cross entropy** directly minimized during training, which is why we refer to perplexity as an *intrinsic metric*.

Perplexity is defined as $2^{H(p,q)/n}$, where $H(p, q)$ is the **cross entropy** between the true distribution of words p and the predicted distribution of words q , and n is the sentence length (the number of words or tokens) used to normalize the score. As cross entropy decreases, perplexity also decreases—the lower the perplexity, the better. While we typically compute cross entropy using a natural logarithm, we calculate cross entropy and perplexity with a base-2 logarithm for the intuitive relationship to hold. (However, whether we use a base-2 or natural logarithm is only a minor implementation detail.)

困惑度，与交叉熵直接相关，因此我们将其称为内生指标。FIXME: 交叉熵损失，不理解 ???

In practice, since the probability for each word in the target sentence is always 1, we compute the cross entropy as the logarithm of the probability scores returned by the model we want to evaluate. In other words, if we have the predicted probability score for each word in a sentence s , we can compute the perplexity directly as follows:



$$\text{Perplexity}(s) = 2^{-\frac{1}{n} \log_2(p(s))}$$

where s is the sentence or text we want to evaluate, such as "The quick brown fox jumps over the lazy dog," $p(s)$ is the probability scores returned by the model, and n is the number of words or tokens. For example, if the model returns the probability scores $[0.99, 0.85, 0.89, 0.99, 0.99, 0.99, 0.99, 0.99]$, the perplexity is:

$$\begin{aligned} & 2^{-\frac{1}{8} \cdot \sum_i \log_2 p(w_i)} \\ &= 2^{-\frac{1}{8} \cdot \sum \log_2(0.99 \times 0.85 \times 0.89 \times 0.99 \times 0.99 \times 0.99 \times 0.99 \times 0.99)} \\ &= 1.043 \end{aligned}$$

If the sentence was "The fast black cat jumps over the lazy dog," with probabilities $[0.99, 0.65, 0.13, 0.05, 0.21, 0.99, 0.99, 0.99]$, the corresponding perplexity would be 2.419.

You can find a code implementation and example of this calculation in the *supplementary/q19-evaluation-llms* subfolder at

<https://github.com/rasbt/MachineLearning-QandAI-book>.

BLEU Score

BLEU is the most popular and most widely used metric for evaluating translated texts. It's used in almost all LLMs capable of translation, including popular tools such as OpenAI's Whisper and GPT models.

BLEU is a reference-based metric that compares the model output to human-generated references and was first developed to capture or automate the essence of human evaluation. In short, BLEU measures the lexical overlap between the model output and the human-generated references based on a precision score.

BLEU 是一个基于精确率的指标，它衡量模型输出与人工生成参考之间的词汇重叠程度。

In more detail, as a precision-based metric, BLEU counts how many words in the generated text (candidate text) occur in the reference text divided by the candidate text length (the number of words), where the reference text is a sample translation provided by a human, for example. This is commonly done for n-grams



rather than individual words, but for simplicity, we will stick to words or 1-grams. (In practice, BLEU is often computed for 4-grams.)

Figure 19.1 demonstrates the BLEU score calculation, using the example of calculating the 1-gram BLEU score. The individual steps in Figure 19.1 illustrate how we compute the 1-gram BLEU score based on its individual components, the weighted precision times a brevity penalty. You can also find a code implementation of this calculation in the [supplementary/q15-text-augment](#) subfolder at <https://github.com/rasbt/MachineLearning-QandAI-book>.

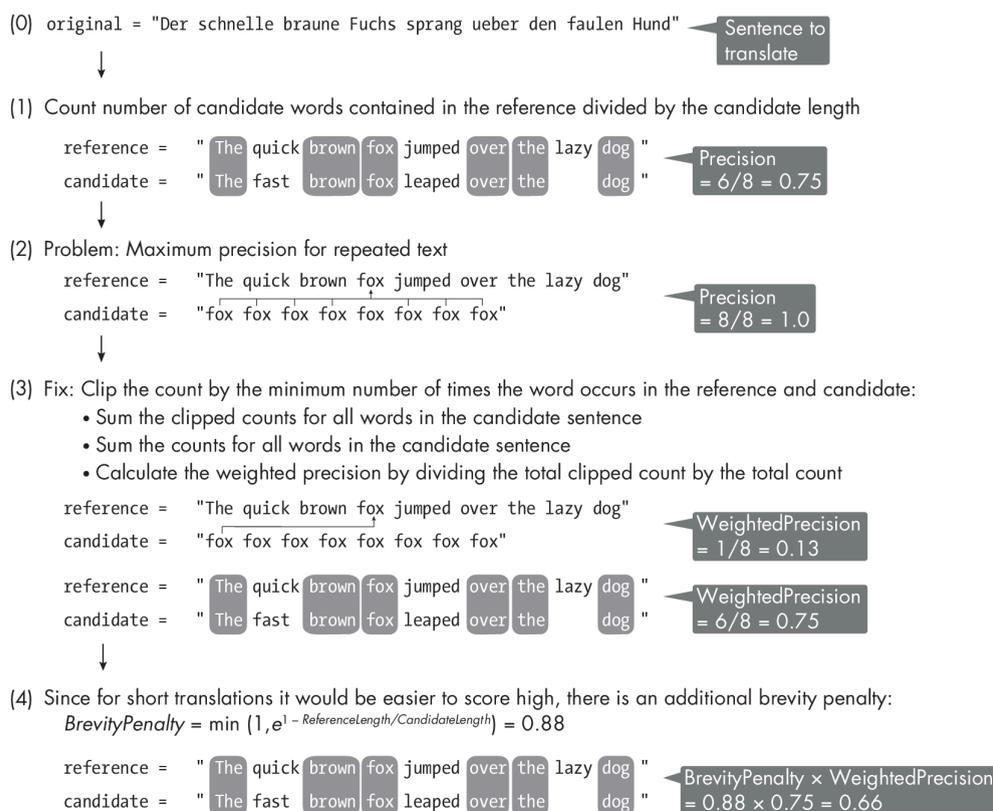


Figure 19.1

BLEU has several shortcomings, mostly owing to the fact that it measures string similarity, and similarity alone is not sufficient for capturing quality. For instance, sentences with similar words but different word orders might still score high, even though altering the word order can significantly change the meaning of a sentence and result in poor grammatical structure. Furthermore, since BLEU relies on exact string matches, it is sensitive to lexical variations and is incapable of identifying semantically similar translations that use synonyms or paraphrases. In other words, BLEU may assign lower scores to translations that are, in fact, accurate and meaningful.



BLEU 有以下几个缺点，主要是由于它衡量的是字符串相似度，而相似度本身不足以捕捉质量。

- 例如，具有相似单词但不同词序的句子可能得分很高，即使改变词序可能会显著改变句子的意义并导致糟糕的语法结构。
- 此外，由于 BLEU 依赖于精确的字符串匹配，它对词汇变体很敏感，无法识别使用同义词或释义的语义相似翻译。
- 换句话说，BLEU 可能会给那些实际上准确且有意义的翻译，分配较低的分。

The original BLEU paper found a high correlation with human evaluations, though this was disproven later.

原始的 BLEU 论文发现与人工评估有很高的相关性，尽管后来被证明是错误的。

Is BLEU flawed? Yes. Is it still useful? Also yes. BLEU is a helpful tool to measure or assess whether a model improves during training, as a proxy for fluency. However, it may not reliably give a correct assessment of the quality of the generated translations and is not well suited for detecting issues. In other words, it's best used as a model selection tool, not a model evaluation tool.

尽管 BLEU 有上述缺点，但它仍然是有用的，可以作为**流畅性**的代理来衡量或评估模型在训练中的改进。然而，它可能无法可靠地评估生成翻译的质量，并且不适合检测问题。换句话说，它最好用作模型选择工具，而不是模型评估工具。

At the time of writing, the most popular alternatives to BLEU are METEOR and COMET (see the References section at the end of this chapter for more details).

目前，最流行的替代 BLEU 的指标是 METEOR 和 COMET（见本章末尾的 "METEOR 和 COMET" 部分，了解更多细节）。

ROUGE Score

While BLEU is commonly used for translation tasks, ROUGE is a popular metric for scoring text summaries.

BLEU 通常用于翻译任务，但 ROUGE 是一个流行的评分文本摘要的指标。

There are many similarities between BLEU and ROUGE. The precision-based BLEU score check shows how many words in the candidate translation occur in the reference translation. The ROUGE score also takes a flipped approach, checking how many words in the reference text appear in the generated text (here typically a summarization instead of a translation); this can be interpreted as a recall-based score.

现代的实现将 ROUGE 计算为 F1 分数，这是召回率（参考文本中出现多少个词）和精确率（候选文本中出现多少个词）的调和平均值。

Modern implementations compute ROUGE as an **F1** score that is the harmonic mean of recall (how many words in the reference occur in the candidate text) and precision (how many words in the candidate text occur in the reference text). For example, Figure 19.2 shows a 1-gram ROUGE score computation (though in practice, ROUGE is often computed for bigrams, that is, 2-grams).

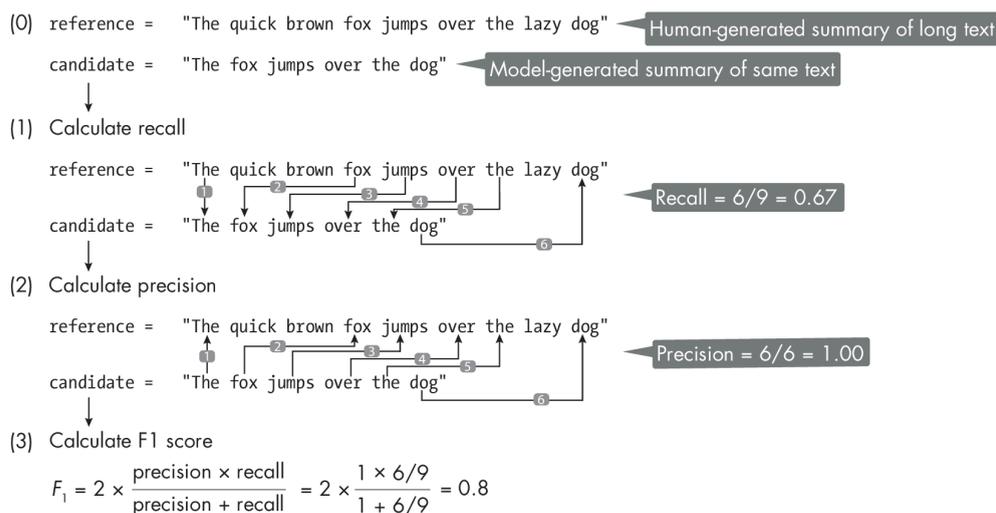


Figure 19.2

There are other ROUGE variants beyond ROUGE-1 (the F1 score--based ROUGE score for 1-grams):



- **ROUGE-N** Measures the overlap of n-grams between the candidate and reference summaries. For example, ROUGE-1 would look at the overlap of individual words (1-grams), while ROUGE-2 would consider the overlap of 2-grams (bigrams).
- **ROUGE-L** Measures the longest common subsequence (LCS) between the candidate and reference summaries. This metric captures the longest co-occurring in-order subsequence of words, which may have gaps in between them.
- **ROUGE-S** Measures the overlap of *skip-bigrams*, or word pairs with a flexible number of words in between them. It can be useful to capture the similarity between sentences with different word orderings.

ROUGE shares similar weaknesses with BLEU. Like BLEU, ROUGE does not account for synonyms or paraphrases. It measures the n-gram overlap between the candidate and reference summaries, which can lead to lower scores for semantically similar but lexically different sentences. However, it's still worth knowing about ROUGE since, according to a study, *all* papers introducing new summarization models at computational linguistics conferences in 2021 used it, and 69 percent of those papers used *only* ROUGE.

ROUGE 与 BLEU 有类似的缺点。与 BLEU 一样，ROUGE 不考虑同义词或释义。它衡量候选文本和参考文本的 n-gram 重叠，这可能导致语义相似但词汇不同的句子的得分较低。然而，了解 ROUGE 仍然是有价值的，因为根据一项研究，2021 年计算语言学会议上介绍的所有新摘要模型都使用了它，其中 69% 的论文只使用了 ROUGE。

BERTScore

Another more recently developed extrinsic metric is **BERTScore** .

For readers familiar with the inception score for generative vision models, BERTScore takes a similar approach, using embeddings from a pretrained model (for more on embeddings, see Chapter [\[ch01\]](#)). Here, BERTScore measures the similarity between a candidate text and a reference text by leveraging the contextual embeddings produced by the BERT model (the encoder-style transformer discussed in Chapter [\[ch17\]](#)).



The steps to compute BERTScore are as follows:

1. Obtain the candidate text via the LLM you want to evaluate (PaLM, LLaMA, GPT, BLOOM, and so on).
2. Tokenize the candidate and reference texts into subwords, preferably using the same tokenizer used for training BERT.
3. Use a pretrained BERT model to create the embeddings for all tokens in the candidate and reference texts.
4. Compare each token embedding in the candidate text to all token embeddings in the reference text, computing their cosine similarity.
5. Align each token in the candidate text with the token in the reference text that has the highest cosine similarity.
6. Compute the final BERTScore by taking the average similarity scores of all tokens in the candidate text.

Figure 19.3 further illustrates these six steps. You can also find a computational example in the `subfolder/q15-text-augment` subfolder at <https://github.com/rasbt/MachineLearning-QandAI-book>.

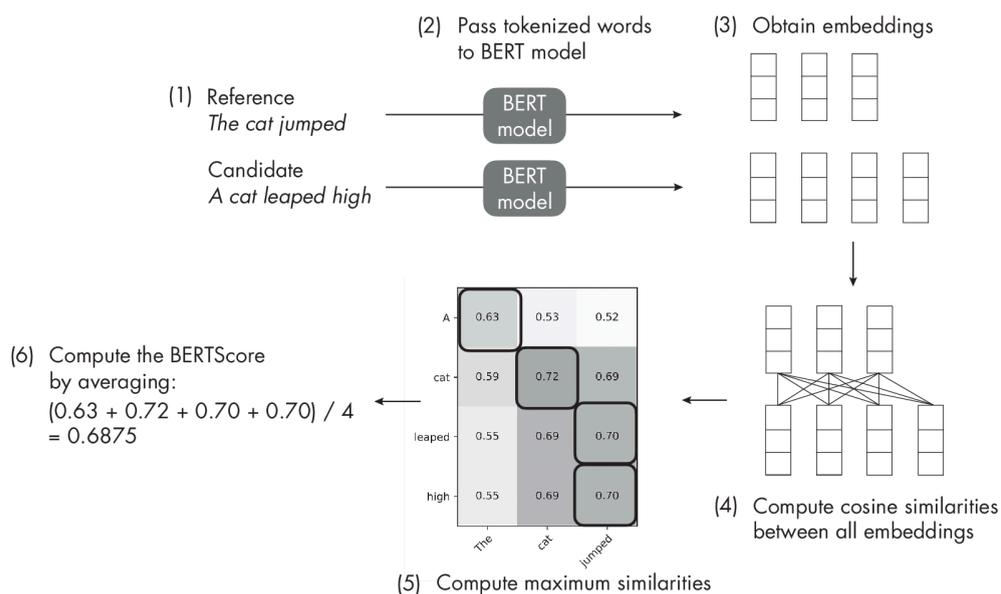


Figure 19.3

BERTScore can be used for translations and summaries, and it captures the semantic similarity better than traditional metrics like BLEU and ROUGE. However,



BERTScore is more robust in paraphrasing than BLEU and ROUGE and captures semantic similarity better due to its contextual embeddings. Also, it may be computationally more expensive than BLEU and ROUGE, as it requires using a pretrained BERT model for the evaluation. While BERTScore provides a useful automatic evaluation metric, it's not perfect and should be used alongside other evaluation techniques, including human judgment.

Surrogate Metrics

All metrics covered in this chapter are surrogates or proxies to evaluate how useful the model is in terms of measuring how well the model compares to human performance for accomplishing a goal. As mentioned earlier, the best way to evaluate LLMs is to assign human raters who judge the results. However, since this is often expensive and not easy to scale, we use the aforementioned metrics to estimate model performance. To quote from the InstructGPT paper "Training Language Models to Follow Instructions with Human Feedback": "Public NLP datasets are not reflective of how our language models are used ... [They] are designed to capture tasks that are easy to evaluate with automatic metrics."

这些指标只是对模型实际表现的近似衡量。正如InstructGPT论文所说，公开的NLP数据集并不能真实反映模型的实际应用场景，它们主要用于便于自动化指标评估的任务。

Besides perplexity, ROUGE, BLEU, and BERTScore, several other popular evaluation metrics are used to assess the predictive performance of LLMs.

Exercises

19-1. In step 5 of Figure [19.3](#), the cosine similarity between the two embeddings of "cat"? is not 1.0, where 1.0 indicates a maximum cosine similarity. Why is that?

19-2. In practice, we might find that the BERTScore is not symmetric. This means that switching the candidate and reference sentences could result in different BERTScores for specific texts. How could we address this?



References

- The paper proposing the original BLEU method: Kishore Papineni et al., "BLEU: A Method for Automatic Evaluation of Machine Translation"? (2002), <https://aclanthology.org/P02-1040/>.
- A follow-up study disproving BLEU's high correlation with human evaluations: Chris Callison-Burch, Miles Osborne, and Philipp Koehn, "Re-Evaluating the Role of BLEU in Machine Translation Research"? (2006), <https://aclanthology.org/E06-1032/>.
- The shortcomings of BLEU, based on 37 studies published over 20 years: Benjamin Marie, "12 Critical Flaws of BLEU"? (2022), https://medium.com/@bnjmn_marie/12-critical-flaws-of-bleu-1d790ccbe1b1.
- The paper proposing the original ROUGE method: Chin-Yew Lin, "ROUGE: A Package for Automatic Evaluation of Summaries"? (2004), <https://aclanthology.org/W04-1013/>.
- A survey on the usage of ROUGE in conference papers: Sebastian Gehrmann, Elizabeth Clark, and Thibault Sellam, "Repairing the Cracked Foundation: A Survey of Obstacles in Evaluation Practices for Generated Text"? (2022), <https://arxiv.org/abs/2202.06935>.
- BERTScore, an evaluation metric based on a large language model: Tianyi Zhang et al., "BERTScore: Evaluating Text Generation with BERT"? (2019), <https://arxiv.org/abs/1904.09675>.
- A comprehensive survey on evaluation metrics for large language models: Asli Celikyilmaz, Elizabeth Clark, and Jianfeng Gao, "Evaluation of Text Generation: A Survey"? (2021), <https://arxiv.org/abs/2006.14799>.
- METEOR is a machine translation metric that improves upon BLEU by using advanced matching techniques and aiming for better correlation with human judgment at the sentence level: Satanjeev Banerjee and Alon Lavie, "METEOR: An Automatic Metric for MT Evaluation with Improved Correlation with Human Judgments"? (2005), <https://aclanthology.org/W05-0909/>.

- COMET is a neural framework that sets new standards for correlating machine translation quality with human judgments, using cross-lingual pretrained models and multiple types of evaluation: Ricardo Rei et al., "COMET: A Neural Framework for MT Evaluation"? (2020), <https://arxiv.org/abs/2009.09025>.
 - The InstructGPT paper: Long Ouyang et al., "Training Language Models to Follow Instructions with Human Feedback"? (2022), <https://arxiv.org/abs/2203.02155>.
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Chapter 20: Stateless and Stateful Training

What is the difference between stateless and stateful training workflows in the context of production and deployment systems?

Stateless training and stateful training refer to different ways of training a production model.

Tips: 无状态训练 `stateless` 和有状态训练 `stateful`，是两种不同的训练方式；训练生产模型时，需要考虑用哪种方式。

Stateless (Re)training

In stateless training, the more conventional approach, we first train an `initial model` on the original training set and then retrain it as new data arrives. Hence, stateless training is also commonly referred to as stateless *retraining*.

Tips: 无状态训练，是先训练一个 `初始模型`，然后在新数据到达时，重新训练模型；可以简单认为是 `树状结构`，初始模型是 `父节点`、衍生出一堆 `叶子节点` 模型。

As Figure [20.1](#) shows, we can think of stateless retraining as a sliding window approach in which we retrain the initial model on different parts of the data from a given data stream.

Tips: 图示中，`初始模型` 是 `父节点`，`新模型` 是 `叶子节点`，`新模型` 是基于 `初始模型` 训练的；训练新模型时，会截取 `滑动窗口` 数据。

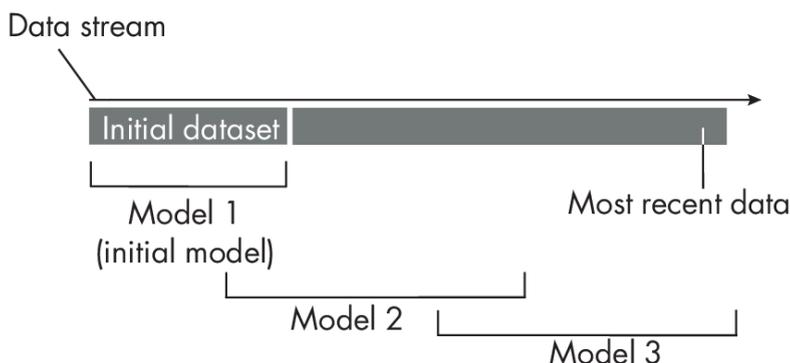


Figure 20.1

For example, to update the initial model in Figure 20.1 (Model 1) to a newer model (Model 2), we train the model on 30 percent of the initial data and 70 percent of the most recent data at a given point in time.

Stateless retraining is a straightforward approach that allows us to adapt the model to the most recent changes in the data and feature-target relationships via retraining the model from scratch in user-defined checkpoint intervals. This approach is prevalent with conventional machine learning systems that cannot be fine-tuned as part of a transfer or self-supervised learning workflow (see Chapter [ch02]).

Tips: 传统的模型，中无状态训练，比较流行，比如 随机森林、梯度提升等，这些都是无法 微调 的

For example, standard implementations of tree-based models, such as random forests and gradient boosting (XGBoost, CatBoost, and LightGBM), fall into this category.

Stateful Training

In stateful training, we train the model on an initial batch of data and then update it periodically (as opposed to retraining it) when new data arrives.

Tips: 有状态的训练，可以认为是 链式结构，初始模型 -> 新模型 -> 新模型 -> ...，每次都基于最新模型叠加而来。



As illustrated in Figure 20.2, we do not retrain the initial model (Model 1.0) from scratch; instead, we update or fine-tune it as new data arrives. This approach is particularly attractive for models compatible with transfer learning or self-supervised learning.



Figure 20.2

The stateful approach mimics a transfer or self-supervised learning workflow where we adopt a pretrained model for fine-tuning. However, stateful training differs fundamentally from transfer and self-supervised learning because it updates the model to accommodate concept, feature, and label drifts. In contrast, transfer and self-supervised learning aim to adopt the model for a different classification task. For instance, in transfer learning, the target labels often differ. In self-supervised learning, we obtain the target labels from the dataset features.

Tips: 有状态的训练，跟 **迁移学习**、**自监督学习**，有本质区别；有状态的训练，会更新模型，以适应概念、特征、标签的漂移；而迁移学习、自监督学习，是基于预训练模型，进行微调。

One significant advantage of stateful training is that we do not need to store data for retraining; instead, we can use it to update the model as soon as it arrives. This is particularly attractive when data storage is a concern due to privacy or resource limitations.

Tips: 有状态的训练，不需要存储数据，可以及时更新模型；这在 **隐私** 或 **资源有限** 的情况下，特别有用。

Exercises

20-1. Suppose we train a classifier for stock trading recommendations using a random forest model, including the moving average of the stock price as a feature. Since new stock market data arrives daily, we are considering how to

update the classifier daily to keep it up to date. Should we take a stateful training or stateless retraining approach to update the classifier?



► Answer, Click to expand

20-2. Suppose we deploy a large language model (transformer) such as ChatGPT that can answer user queries. The dialogue interface includes thumbs-up and thumbs-down buttons so that users can give direct feedback based on the generated queries. While collecting the user feedback, we don't update the model immediately as new feedback arrives. However, we are planning to release a new or updated model at least once per month. Should we use stateless or stateful retraining for this model?



Chapter 21: Data-Centric AI

What is data-centric AI, how does it compare to the conventional modeling paradigm, and how do we decide whether it's the right fit for a project?

Data-centric AI is a paradigm or workflow in which we keep the model training procedure fixed and iterate over the dataset to improve the predictive performance of a model. The following sections define what data-centric AI means in more detail and compare it to conventional model-centric approaches.

Tips: **数据驱动 AI** 是一种范式或工作流，其中我们保持模型训练过程不变，通过迭代**数据集**来提高模型的预测性能。

Data-Centric vs. Model-Centric AI

In the context of data-centric AI, we can think of the conventional workflow, which is often part of academic publishing, as model-centric AI. However, in an academic research setting, we are typically interested in developing new methods (for example, neural network architectures or loss functions). Here, we consider existing benchmark datasets to compare the new method to previous approaches and determine whether it is an improvement over the status quo.

Tips: 在 **数据驱动 AI** 的上下文中，我们可以将传统的学术优化的模型视为 **模型驱动 AI**。

- 在学术研究环境中，我们通常对开发**新方法**（例如神经网络架构或损失函数）感兴趣。
- 我们考虑现有的**基准数据集**，以比较新方法与先前的方法，并确定它是否更优。

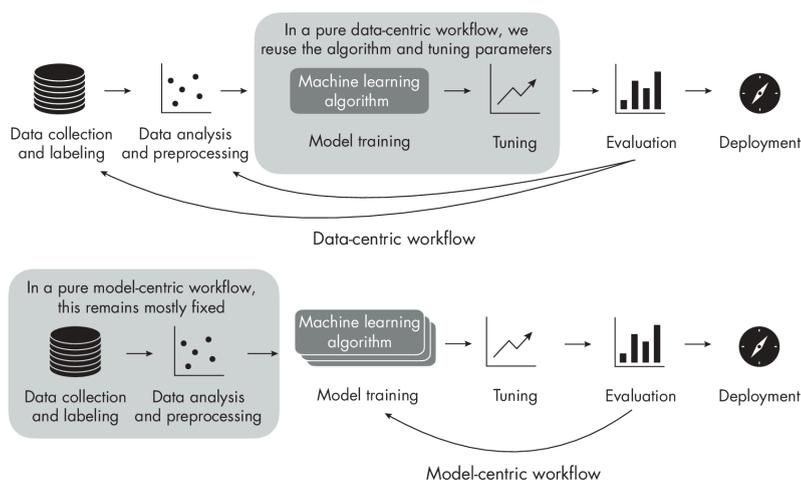


Figure 21.1

While *data-centric AI* is a relatively new term, the idea behind it is not. Many people I've spoken with say they used a data-centric approach in their projects before the term was coined. In my opinion, data-centric AI was created to make "caring about data quality" attractive again, as data collection and curation are often considered tedious or thankless. This is analogous to how the term *deep learning* made neural networks interesting again in the early 2010s.

Tips: 虽然 **数据驱动 AI** 是一个相对较新的术语，但其背后的理念并不新。

- 许多人告诉我，在他们使用 **数据驱动 AI** 之前，他们已经在项目中使用了 **数据驱动 AI**。
- 在我看来，**数据驱动 AI** 的称呼，是为了让 **关注数据质量** 再次变得有吸引力，因为数据收集和整理通常被认为是很繁琐或无意义的。
- 这类似于 **深度学习** 在2010年代初期让神经网络变得有趣起来。

Do we need to choose between data-centric and model-centric AI, or can we rely on both? In short, data-centric AI focuses on changing the data to improve performance, while model-centric approaches focus on modifying the model to improve performance. Ideally, we should use both in an applied setting where we want to get the best possible predictive performance. However, in a research setting or an exploratory stage of an applied project, working with too many variables simultaneously is messy. If we change both model and data at once, it's hard to pinpoint which change is responsible for the improvement.

Tips: 我们是否需要在 **数据驱动 AI** 和 **模型驱动 AI** 之间做出选择，或者我们可以依赖两者？

- 简而言之，**数据驱动 AI** 专注于**改变数据**以提高性能，而 **模型驱动 AI** 专注于**修改模型**以提高性能。
- 理想情况下，我们应该同时使用两者。



It is important to emphasize that data-centric AI is a paradigm and workflow, not a particular technique. Data-centric AI therefore implicitly includes the following:

- Analyses and modifications of training data, from outlier removal to missing data imputation
- Data synthesis and data augmentation techniques
- Data labeling and label-cleaning methods
- The classic active learning setting where a model suggests which data points to label

Tips: 重要的是要强调，**数据驱动 AI** 是一种范式和流程，而不是一种特定的技术，隐含地包括以下内容：

- 训练数据的分析和修改，从异常值去除到缺失数据插补
- 数据合成和数据增强技术
- 数据标注和标签清理方法
- 经典主动学习设置，其中模型建议哪些数据点需要标注

We consider an approach *data centric* if we change only the data (using the methods listed here), not the other aspects of the modeling pipeline.

In machine learning and AI, we often use the phrase "garbage in, garbage out" meaning that poor-quality data will result in a poor predictive model. In other words, we cannot expect a well-performing model from a low-quality dataset.

I've observed a common pattern in applied academic projects that attempt to use machine learning to replace an existing methodology. Often, researchers have only a small dataset of examples (say, hundreds of training examples). Labeling data is often expensive or considered boring and thus best avoided. In these cases, the researchers spend an unreasonable amount of time trying out different machine-learning algorithms and model tuning. To resolve this issue, investing additional time or resources in labeling additional data would be worthwhile.



Tips:

- 通常，研究人员，只有一个 **小数据集**（例如，几百个训练样本）。
- **标注数据**通常很 **昂贵** 或被认为很 **无聊**，因此最好避免。
- 在这些情况下，研究人员**花费了不合理的时间**来尝试不同的机器学习算法和模型调优。
- 为了解决这个问题，投资额外的时间或资源来 **标注更多的数据** 将是有价值的。

The main advantage of data-centric AI is that it puts the data first so that if we invest resources to create a higher-quality dataset, all modeling approaches will benefit from it downstream.

Tips: 数据驱动 AI 的主要优势在于，它将数据放在首位，因此如果我们投资资源来创建更高质量的数据集，所有建模方法都将从中受益。

Recommendations

Taking a data-centric approach is often a good idea in an applied project where we want to improve the predictive performance to solve a particular problem. In this context, it makes sense to start with a modeling baseline and improve the dataset since it's often more worthwhile than trying out bigger, more expensive models.

If our task is to develop a new or better methodology, such as a new neural network architecture or loss function, a model-centric approach might be a better choice. Using an established benchmark dataset without changing it makes it easier to compare the new modeling approach to previous work. Increasing the model size usually improves performance, but so does the addition of training examples. Assuming small training sets ($< 2k$) for classification, extractive question answering, and multiple-choice tasks, adding a hundred examples can result in the same performance gain as adding billions of parameters.

In a real-world project, alternating between data-centric and model-centric modes makes a lot of sense. Investing in data quality early on will benefit all models. Once a good dataset is available, we can begin to focus on model tuning to improve performance.

Tips: 在实际项目中，交替使用 **数据驱动** 和 **模型驱动** 模式是有意义的。

- 早期投资于 **数据质量** 将使所有模型受益。
- 一旦有了一个好的数据集，我们可以开始专注于 **模型调优** 以提高性能。



Exercises

21-1. A recent trend is the increased use of predictive analytics in healthcare. For example, suppose a healthcare provider develops an AI system that analyzes patients' electronic health records and provides recommendations for lifestyle changes or preventive measures. For this, the provider requires patients to monitor and share their health data (such as pulse and blood pressure) daily. Is this an example of data-centric AI?

21-2. Suppose we train a ResNet-34 convolutional neural network to classify images in the CIFAR-10 and ImageNet datasets. To reduce overfitting and improve classification accuracy, we experiment with data augmentation techniques such as image rotation and cropping. Is this approach data centric?

References

- An example of how adding more training data can benefit model performance more than an increase in model size: Yuval Kirstain et al., "A Few More Examples May Be Worth Billions of Parameters"? (2021), <https://arxiv.org/abs/2110.04374>.
- Cleanlab is an open source library that includes methods for improving labeling errors and data quality in computer vision and natural language processing contexts: <https://github.com/cleanlab/cleanlab>.

Chapter 22: Speeding Up Inference



What are techniques to speed up model inference through optimization without changing the model architecture or sacrificing accuracy?

In machine learning and AI, *model inference* refers to making predictions or generating outputs using a trained model. The main general techniques for improving model performance during inference include parallelization, vectorization, loop tiling, operator fusion, and quantization, which are discussed in detail in the following sections.

Tips:

- 优化模型**推理速度**，有多种方法，包括：**并行化**、**向量化**、**循环分块**、**算子融合**、**量化**等。
- 这些方法，将在后续章节中详细讨论。

Parallelization

One common way to achieve better parallelization during inference is to run the model on a batch of samples rather than on a single sample at a time. This is sometimes also referred to as *batched inference* and assumes that we are receiving multiple input samples or user inputs simultaneously or within a short time window, as illustrated in Figure [22.1](#).

Tips: **并行化**，也被称为**批量推理**，同时或短时间窗口内，接收到多个输入样本或用户输入，模型同时处理。

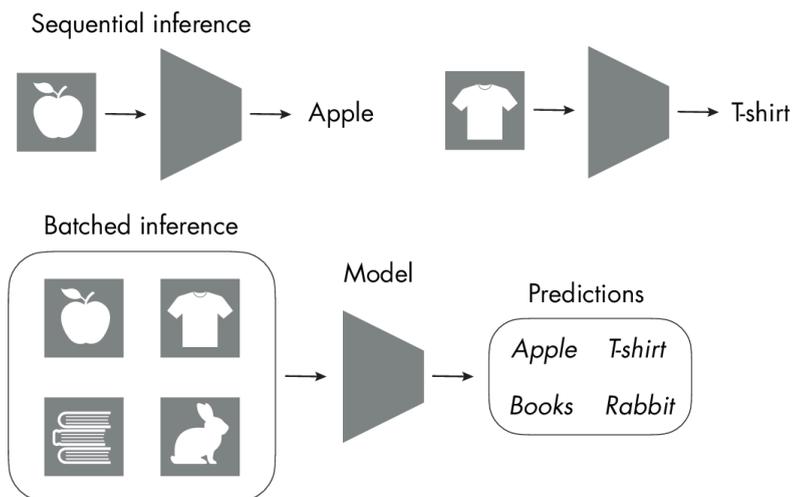


Figure 22.1

Figure [22.1](#) shows sequential inference processing one item at a time, which creates a bottleneck if there are several samples waiting to be classified. In batched inference, the model processes all four samples at the same time.

Vectorization

Vectorization refers to performing operations on entire data structures, such as arrays (tensors) or matrices, in a single step rather than using iterative constructs like `for` loops. Using vectorization, multiple operations from the loop are performed simultaneously using single instruction, multiple data (SIMD) processing, which is available on most modern CPUs.

Tips: 向量化，也被称为 **单指令多数据**，在现代 CPU 上，可以同时处理多个数据。

This approach takes advantage of the low-level optimizations in many computing systems and often results in significant speedups. For example, it might rely on BLAS.

BLAS (which is short for *Basic Linear Algebra Subprograms*) is a specification that prescribes a set of low-level routines for performing common linear algebra operations such as vector addition, scalar multiplication, dot products, matrix multiplication, and others. Many array and deep learning libraries like NumPy and PyTorch use BLAS under the hood.



To illustrate vectorization with an example, suppose we wanted to compute the dot product between two vectors. The non-vectorized way of doing this would be to use a `for` loop, iterating over each element of the array one by one. However, this can be quite slow, especially for large arrays. With vectorization, you can perform the dot product operation on the entire array at once, as shown in Figure 22.2.

Classic for loop	<pre>x = [1.2, 2.2, 3.3, 4.4] w = [5.5, 6.6, 7.7, 8.8] output = 0. for x_j, w_j in zip(x, w): output += x_j * w_j print(output) 85.25</pre>
Vectorized implementation	<pre>import torch x = torch.tensor([1.2, 2.2, 3.3, 4.4]) w = torch.tensor([5.5, 6.6, 7.7, 8.8]) x.dot(w) tensor(85.2500)</pre>

Figure 22.2

In the context of linear algebra or deep learning frameworks like TensorFlow and PyTorch, vectorization is typically done automatically. This is because these frameworks are designed to work with multidimensional arrays (also known as *tensors*), and their operations are inherently vectorized. This means that when you perform functions using these frameworks, you automatically leverage the power of vectorization, resulting in faster and more efficient computations.

Loop Tiling

Loop tiling (also often referred to as *loop nest optimization*) is an advanced optimization technique to enhance data locality by breaking down a loop's iteration space into smaller chunks or "tiles." This ensures that once data is loaded into cache, all possible computations are performed on it before the cache is cleared.

Tips: 循环分块，也被称为 **循环嵌套优化**，将循环的迭代空间分成小块，确保数据加载到缓存后，所有可能的计算都在缓存中完成，然后缓存被清除。



Figure 22.3 illustrates the concept of loop tiling for accessing elements in a two-dimensional array. In a regular `for` loop, we iterate over columns and rows one element at a time, whereas in loop tiling, we subdivide the array into smaller tiles.

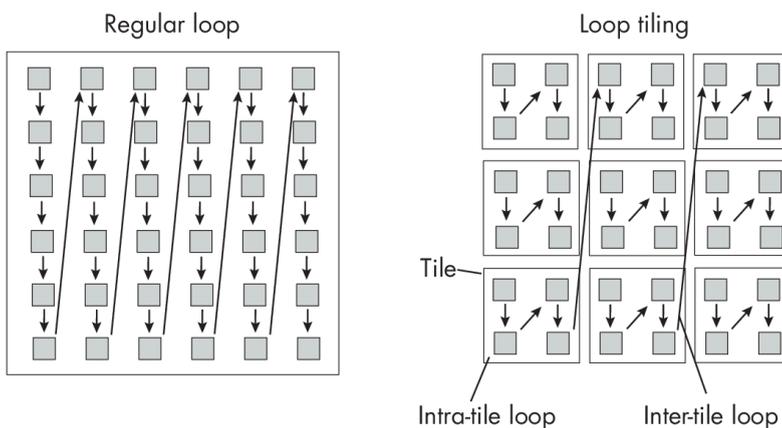


Figure 22.3

Note that in languages such as Python, we don't usually perform loop tiling, because Python and many other high-level languages do not allow control over cache memory like lower-level languages such as C and C++ do. These kinds of optimizations are often handled by underlying libraries like NumPy and PyTorch when performing operations on large arrays.

Tips: 在 Python 等高级语言中，通常不进行循环分块，因为这些语言不提供对缓存内存的控制，如 C 和 C++ 等底层语言。这些优化通常由底层库（如 NumPy 和 PyTorch）在处理大型数组时自动处理。

Operator Fusion

Operator fusion, sometimes called *loop fusion*, is an optimization technique that combines multiple loops into a single loop. This is illustrated in Figure 22.4, the product of an array of numbers are fused into a single loop.

Tips: 算子融合，也被称为 **循环融合**，将多个循环合并成一个循环。



```
numbers = [1, 2, 3, 4, 5]

# First loop to calculate the sum
total_sum = 0
for num in numbers:
    total_sum += num

# Second loop to calculate the product
product = 1
for num in numbers:
    product *= num

print("Sum:", total_sum)
print("Product:", product)

Sum: 15
Product: 120
```

```
numbers = [1, 2, 3, 4, 5]

# Single loop to calculate both
# the sum AND the product
total_sum = 0
product = 1
for num in numbers:
    total_sum += num
    product *= num

print("Sum:", total_sum)
print("Product:", product)

Sum: 15
Product: 120
```

Figure 22.4

Operator fusion can improve the performance of a model by reducing the overhead of loop control, decreasing memory access times by improving cache performance, and possibly enabling further optimizations through vectorization. You might think this behavior of **vectorization** would be incompatible with **loop tiling**, in which we break a **for** loop into multiple loops.

Tips: 算子融合，可以提高模型性能，通过减少循环控制的开销，提高缓存性能，并可能通过向量化进一步优化。

However, these techniques are actually complementary, used for different optimizations, and applicable in different situations. **Operator fusion** is about reducing the total number of loop iterations and improving data locality when the entire data fits into cache. **Loop tiling** is about improving cache utilization when dealing with larger multidimensional arrays that do not fit into cache.

Related to operator fusion is the concept of *reparameterization*, which can often also be used to simplify multiple operations into one. Popular examples include training a network with multibranch architectures that are reparameterized into single-stream architectures during inference. This reparameterization approach differs from traditional operator fusion in that it does not merge multiple operations into a single operation. Instead, it rearranges the operations in the network to create a more efficient architecture for inference. In the so-called RepVGG architecture, for example, each branch during training consists of a series of convolutions. Once training is complete, the model is reparameterized into a single sequence of convolutions.

Tips: 重参数化，也被称为 **重参数化优化**，将多个操作合并成一个操作。



Quantization

Quantization reduces the computational and storage requirements of machine learning models, particularly deep neural networks. This technique involves converting the floating-point numbers (technically discrete but representing continuous values within a specific range) for implementing weights and biases in a trained neural network to more discrete, lower-precision representations such as integers. Using less precision reduces the model size and makes it quicker to execute, which can lead to significant improvements in speed and hardware efficiency during inference.

Tips: 量化，也被称为 **量化优化**，将浮点数转换为整数，减少模型大小和计算量，提高推理速度。

In the realm of deep learning, it has become increasingly common to quantize trained models down to 8-bit and 4-bit integers. These techniques are especially prevalent in the deployment of large language models.

There are two main categories of quantization. In **post-training quantization**, the model is first trained normally with full-precision weights, which are then quantized after training. **Quantization-aware training**, on the other hand, introduces the quantization step during the training process. This allows the model to learn to compensate for the effects of quantization, which can help maintain the model's accuracy.

Tips: 量化，一般分为 2 大类：**后训练量化**、**量化感知训练**。

- 后训练量化，在训练完成后，对模型进行量化。
- 量化感知训练，在训练过程中，引入量化步骤，让模型学习量化带来的影响。

However, it's important to note that quantization can occasionally lead to a reduction in model accuracy. Since this chapter focuses on techniques to speed up model inference *without* sacrificing accuracy, quantization is not as good a fit for this chapter as the previous categories.

Tips: 量化，可能会导致模型精度下降，因此，本章再不讨论量化。



Other techniques to improve inference speeds include knowledge distillation and pruning, discussed in Chapter [ch06]. However, these techniques affect the model architecture, resulting in smaller models, so they are out of scope for this chapter's question.

Tips: 其他提升推理速度的策略，包括：知识蒸馏、剪枝等，之前章节已经讨论过；但是，这些策略会影响模型架构，导致模型变小，因此，也不在本章讨论范围内。

Exercises

22-1. Chapter [ch07] covered several multi-GPU training paradigms to speed up model training. Using multiple GPUs can, in theory, also speed up model inference. However, in reality, this approach is often not the most efficient or most practical option. Why is that?

22-2. Vectorization and loop tiling are two strategies for optimizing operations that involve accessing array elements. What would be the ideal situation in which to use each?

References

- The official BLAS website: <https://www.netlib.org/blas/>.
- The paper that proposed loop tiling: Michael Wolfe, "More Iteration Space Tiling"? (1989), <https://dl.acm.org/doi/abs/10.1145/76263.76337>.
- RepVGG CNN architecture merging operations in inference mode: Xiaohan Ding et al., "RepVGG: Making VGG-style ConvNets Great Again"? (2021), <https://arxiv.org/abs/2101.03697>.
- A new method for quantizing the weights in large language models down to 8-bit integer representations: Tim Dettmers et al., "LLM.int8(): 8-bit Matrix Multiplication for Transformers at Scale"? (2022), <https://arxiv.org/abs/2208.07339>.

- A new method for quantizing the weights in LLMs farther down to 4-bit integers: Elias Frantar et al., "GPTQ: Accurate Post-Training Quantization for Generative Pre-trained Transformers"? (2022), <https://arxiv.org/abs/2210.17323>.
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Chapter 23: Data Distribution Shifts

What are the main types of data distribution shifts we may encounter after model deployment?

Data distribution shifts are one of the most common problems when putting machine learning and AI models into production. In short, they refer to the differences between the distribution of data on which a model was trained and the distribution of data it encounters in the real world. Often, these changes can lead to significant drops in model performance because the model's predictions are no longer accurate.

Tips:数据分布偏移是生产环境中使用模型时，最常见的问题。

- 指的是，模型在训练时所使用的数据分布，与在实际应用中遇到的数据分布之间的差异。
- 通常，这些变化会导致模型性能显著下降，因为模型的预测不再准确。

There are several types of distribution shifts, some of which are more problematic than others. The most common are covariate shift, concept drift, label shift, and domain shift; all discussed in more detail in the following sections.

Tips:

- 数据分布偏移，有多种类型，其中最常见的是：协变量偏移、概念漂移、标签偏移和域偏移。
- 这些偏移类型，将在后续章节中详细讨论。

Covariate Shift

Suppose $p(x)$ describes the distribution of the input data (for instance, the features), $p(y)$ refers to the distribution of the target variable (or class label distribution), and $p(y|x)$ is the distribution of the targets y given the inputs x .

Covariate shift happens when the distribution of the input data, $p(x)$, changes, but the conditional distribution of the output given the input, $p(y|x)$, remains the same.



Tips:

- 协变量偏移，指的是，输入数据分布 $p(x)$ 发生变化，但输出条件分布 $p(y|x)$ 保持不变。
- 协变量 covariate，一般是指特征变量，通常会影响输出结果，但并不一定是主要因素。

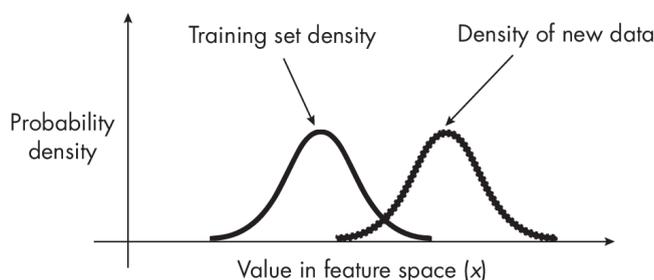


Figure 23.1

For example, suppose we trained a model to predict whether an email is spam based on specific features. Now, after we embed the email spam filter in an email client, the email messages that customers receive have drastically different features. For example, the email messages are much longer and are sent from someone in a different time zone. However, if the way those features relate to an email being spam or not doesn't change, then we have a covariate shift.

Covariate shift is a very common challenge when deploying machine learning models. It means that the data the model receives in a live or production environment is different from the data on which it was trained. However, because the relationship between inputs and outputs, $p(y|x)$, remains the same under covariate shift, techniques are available to adjust for it.

Tips:

- 协变量偏移，模型在实际应用中遇到的**数据分布**，与在训练时所使用的数据分布**不同**。
- 但是，由于输入和输出之间的关系 $p(y|x)$ 保持不变，因此有调整方法，例如：对抗验证、重要性加权等。

A common technique to detect covariate shift is *adversarial validation*, which is covered in more detail in Chapter [\[ch29\]](#). Once covariate shift is detected, a common method to deal with it is **importance weighting**, which assigns different



weights to the training example to emphasize or de-emphasize certain instances during training. Essentially, instances that are more likely to appear in the test distribution are given more weight, while instances that are less likely to occur are given less weight. This approach allows the model to focus more on the instances representative of the test data during training, making it more robust to covariate shift.

Label Shift

Label shift, sometimes referred to as *prior probability shift*, occurs when the class label distribution $p(y)$ changes, but the class-conditional distribution $p(y|x)$ remains unchanged. In other words, there is a significant change in the label distribution or target variable.

Tips:

- 标签偏移，指的是，标签分布 $p(y)$ 发生变化，但条件分布 $p(y|x)$ 保持不变。
- 标签偏移，通常与目标变量（或类标签分布）的变化有关。

As an example of such a scenario, suppose we trained an email spam classifier on a balanced training dataset with 50 percent spam and 50 percent non-spam email. In contrast, in the real world, only 10 percent of email messages are spam.

A common way to address label shifts is to update the model using the **weighted loss function**, especially when we have an idea of the new distribution of the labels. This is essentially a form of **importance weighting**. By adjusting the weights in the loss function according to the new label distribution, we are incentivizing the model to pay more attention to certain classes that have become more common (or less common) in the new data. This helps align the model's predictions more closely with the current reality, improving its performance on the new data.

Tips: 损失函数加权，突出重要样本分类。

Concept Drift



Concept drift refers to the change in the mapping between the input features and the target variable. In other words, concept drift is typically associated with changes in the conditional distribution $p(y|x)$, such as the relationship between the inputs x and the output y .

Tips:

- 概念漂移，指的是，输入特征与目标变量之间的映射关系发生变化。
- 概念漂移，通常与**条件分布** $p(y|x)$ 的变化有关。

Using the example of the spam email classifier from the previous section, the features of the email messages might remain the same, but *how* those features relate to whether an email is spam might change. This could be due to a new spamming strategy that wasn't present in the training data. Concept drift can be much harder to deal with than the other distribution shifts discussed so far since it requires continuous monitoring and potential model retraining.

Domain Shift

The terms *domain shift* and *concept drift* are used somewhat inconsistently across the literature and are sometimes taken to be interchangeable. In reality, the two are related but slightly different phenomena. *Concept drift* refers to a change in the function that maps from the inputs to the outputs, specifically to situations where the relationship between features and target variables changes as more data is collected over time.

Tips:

- 领域偏移，通常跟概念漂移有差异。
- 领域偏移，指的是，输入数据分布 $p(x)$ 和输出条件分布 $p(y|x)$ 都发生变化。
- 领域偏移，也被称为**联合分布偏移**，因为联合分布 $p(x, y)$ 是输入和输出分布的乘积。

In *domain shift*, the distribution of inputs, $p(x)$, and the conditional distribution of outputs given inputs, $p(y|x)$, both change. This is sometimes also called *joint distribution shift* due to the joint distribution:



$$p(x, y) = p(y|x) \cdot p(x)$$

We can thus think of domain shift as a combination of both covariate shift and concept drift. In addition, since we can obtain the marginal distribution $p(y)$ by integrating over the joint distribution $p(x, y)$ over the variable x (mathematically expressed as $p(y) = \int p(x, y) dx$), covariate drift and concept shift also imply label shift. (However, exceptions may exist where the change in $p(x)$ compensates for the change in $p(y|x)$ such that $p(y)$ may not change.) Conversely, label shift and concept drift usually also imply covariate shift.

To return once more to the example of email spam classification, domain shift would mean that the features (content and structure of email) *and* the relationship between the features and target both change over time. For instance, spam email in 2023 might have different features (new types of phishing schemes, new language, and so forth), and the definition of what constitutes spam might have changed as well. This type of shift would be the most challenging scenario for a spam filter trained on 2020 data, as it would have to adjust to changes in both the input data and the target concept.

Domain shift is perhaps the most difficult type of shift to handle, but monitoring model performance and data statistics over time can help detect domain shifts early. Once they are detected, mitigation strategies include collecting more labeled data from the target domain and retraining or adapting the model.

Types of Data Distribution Shifts

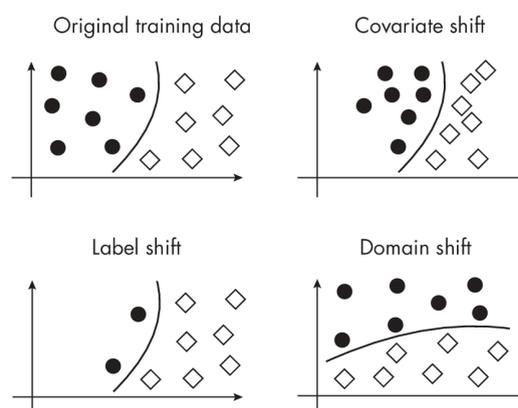


Figure 23.2

As noted in the previous sections, some types of distribution shift are more problematic than others. The least problematic among them is typically



covariate shift. Here, the distribution of the input features, $p(x)$, changes between the training and testing data, but the conditional distribution of the output given the inputs, $p(y|x)$, remains constant. Since the underlying relationship between the inputs and outputs remains the same, the model trained on the training data can still apply, in principle, to the testing data and new data.

The most problematic type of distribution shift is typically **joint distribution shift**, where both the input distribution $p(x)$ and the conditional output distribution $p(y|x)$ change. This makes it particularly difficult for a model to adjust, as the learned relationship from the training data may no longer hold. The model has to cope with both new input patterns and new rules for making predictions based on those patterns.

However, the "severity" of a shift can vary widely depending on the real-world context. For example, even a covariate shift can be extremely problematic if the shift is severe or if the model cannot adapt to the new input distribution. On the other hand, a joint distribution shift might be manageable if the shift is relatively minor or if we have access to a sufficient amount of labeled data from the new distribution to retrain our model.

In general, it's crucial to monitor our models' performance and be aware of potential shifts in the data distribution so that we can take appropriate action if necessary.

Tips: 监控模型性能，及时发现潜在 **数据分布偏移**，非常重要。

Exercises

23-1. What is the big issue with importance weighting as a technique to mitigate covariate shift?

23-2. How can we detect these types of shifts in real-world scenarios, especially when we do not have access to labels for the new data?

References

- Recommendations and pointers to advanced mitigation techniques for avoiding domain shift: Abolfazl Farahani et al., "A Brief Review of Domain Adaptation"? (2020), <https://arxiv.org/abs/2010.03978>.
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Chapter 24: Poisson and Ordinal Regression

When is it preferable to use **Poisson regression** over **Ordinal regression**, and vice versa?

本章讨论了两种回归模型：泊松回归和序数回归，并讨论了它们的应用场景。

- 泊松回归用于**计数数据**，序数回归用于**有序数据**。

We usually use **Poisson regression** when the target variable represents count data (**positive integers**). As an example of count data, consider the number of colds contracted on an airplane or the number of guests visiting a restaurant on a given day. Besides the target variable representing counts, the data should also be Poisson distributed, which means that the mean and variance are roughly the same. (For large means, we can use a normal distribution to approximate a Poisson distribution.)

泊松回归通常用于表示计数数据（正整数）的目标变量。例如，考虑飞机上感冒的人数或某天餐厅的客人数量。除了表示计数的目标变量外，数据还应服从泊松分布，这意味着均值和方差大致相同。（对于大均值，我们可以使用正态分布来近似泊松分布。）

更多细节：[泊松分布](#)

Ordinal data is a subcategory of categorical data where the categories have a natural order, such as $1 < 2 < 3$, as illustrated in Figure [24.1](#). Ordinal data is often represented as positive integers and may look similar to count data. For example, consider the star rating on Amazon (1 star, 2 stars, 3 stars, and so on). However, ordinal regression does not make any assumptions about the distance between the ordered categories. Consider the following measure of disease severity: *severe* > *moderate* > *mild* > *none*. While we would typically map the disease severity variable to an integer representation ($4 > 3 > 2 > 1$), there is no assumption that the distance between 4 and 3 (severe and moderate) is the same as the distance between 2 and 1 (mild and none).

序数数据是分类数据的一个子类别，其中类别具有自然顺序，例如 $1 < 2 < 3$ ，如图 1.1 所示。序数数据通常表示为正整数，可能与计数数据相似。例如，考虑亚马逊上的星级评分（1 星、2 星、3 星等）。然而，序数回归对有序类别之间的距离没有任何假设。考虑以下疾病严重程度的衡量标准：*严重* $>$ *中等* $>$ *轻微* $>$ *无*。虽然我们通常将疾病严重程度变量映射为整数表示（ $4 > 3 > 2 > 1$ ），但没有任何假设认为 4 和 3（严重和中等）之间的距离与 2 和 1（轻微和无）之间的距离相同。

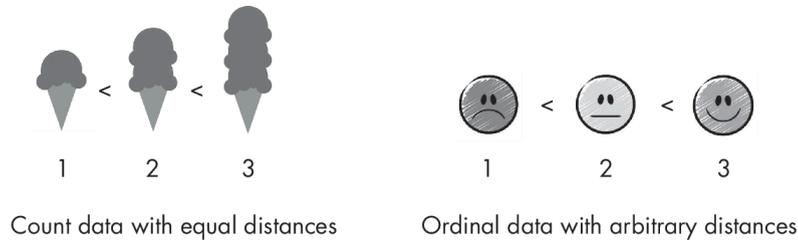


Figure 24.1

In short, we use Poisson regression for count data. We use Ordinal regression when we know that certain outcomes are "higher" or "lower" than others, but we are not sure how much or if it even matters.

Exercises

24-1. Suppose we want to predict the number of goals a soccer player will score in a particular season. Should we solve this problem using ordinal regression or Poisson regression?

24-2. Suppose we ask someone to sort the last three movies they have watched based on their order of preference. Ignoring the fact that this dataset is a tad too small for machine learning, which approach would be best suited for this kind of data?



Chapter 25: Confidence Intervals

What are the different ways to construct confidence intervals for machine learning classifiers?

There are several ways to construct **confidence intervals** for machine learning models, depending on the model type and the nature of your data. For instance, some methods are computationally expensive when working with deep neural networks and are thus more suitable to less resource-intensive machine learning models. Others require larger datasets to be reliable.

The following are the most common methods for constructing confidence intervals:

- Constructing normal approximation intervals based on a test set
- Bootstrapping training sets
- Bootstrapping the test set predictions
- Confidence intervals from retraining models with different random seeds

Before reviewing these in greater depth, let's briefly review the definition and interpretation of confidence intervals.

Defining Confidence Intervals

A **confidence interval** is a type of method to estimate an unknown population parameter. A **population parameter** is a specific measure of a statistical population, for example, a mean (average) value or proportion. By "specific"? measure, I mean there is a single, exact value for that parameter for the entire population. Even though this value may not be known and often needs to be estimated from a sample, it is a fixed and definite characteristic of the population. A *statistical population*, in turn, is the complete set of items or individuals we study.

置信区间，通常用于估计总体参数的精确度。它提供了一个范围，在这个范围内，我们可以对总体参数的值有较高的信心。

统计学中，**总体参数**是指描述总体分布特征的数值，例如总体均值、总体方差等。

总体参数是固定的、确定的，但通常无法直接观测到，需要通过样本数据进行估计。



In a machine learning context, the population could be considered the entire possible set of instances or data points that the model may encounter, and the parameter we are often most interested in is the true generalization accuracy of our model on this population.

The accuracy we measure on the test set estimates the true generalization accuracy. However, it's subject to random error due to the specific sample of test instances we happened to use. This is where the concept of a confidence interval comes in. A 95 percent confidence interval for the generalization accuracy gives us a range in which we can be reasonably sure that the true generalization accuracy lies.

For instance, if we take 100 different data samples and compute a 95 percent confidence interval for each sample, approximately 95 of the 100 confidence intervals will contain the true population value (such as the generalization accuracy), as illustrated in Figure 25.1.

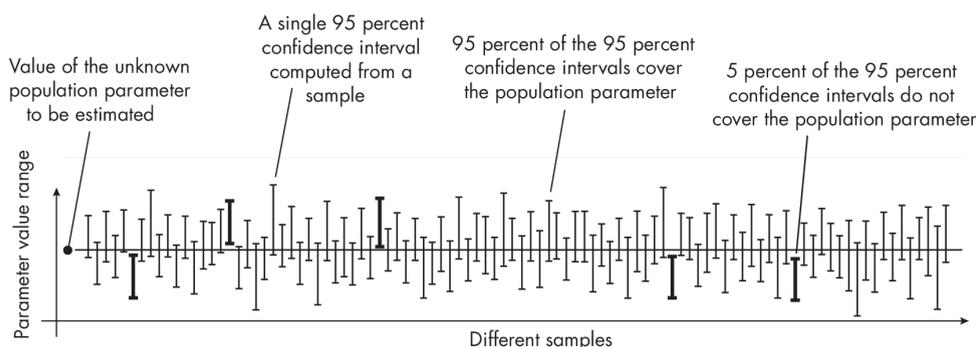


Figure 25.1

More concretely, if we were to draw 100 different representative test sets from the population (for instance, the entire possible set of instances that the model may encounter) and compute the 95 percent confidence interval for the generalization accuracy from each test set, we would expect about 95 of these intervals to contain the true generalization accuracy.



We can display confidence intervals in several ways. It is common to use a bar plot representation where the top of the bar represents the parameter value (for example, model accuracy) and the whiskers denote the upper and lower levels of the confidence interval (left chart of Figure 25.2). Alternatively, the confidence intervals can be shown without bars, as in the right chart of Figure 25.2.

图 1.1 展示了置信区间的两种常见表示方式。左图使用条形图表示，条形的高度表示参数值（例如模型准确率），而须状物则表示置信区间的上下界。右图则不使用条形，直接显示置信区间的上下界。

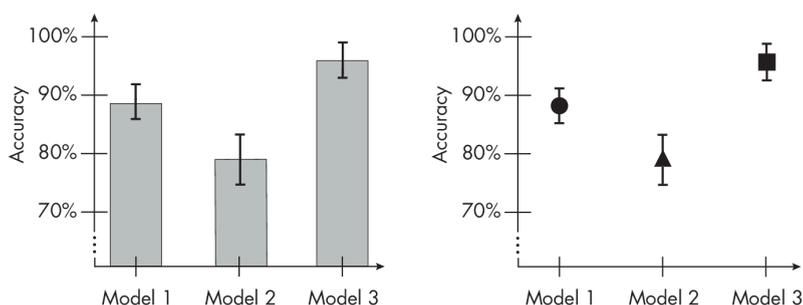


Figure 25.2

This visualization is functionally useful in a number of ways. For instance, when confidence intervals for two model performances do *not* overlap, it's a strong visual indicator that the performances are significantly different. Take the example of statistical significance tests, such as t-tests: if two 95 percent confidence intervals do not overlap, it strongly suggests that the difference between the two measurements is statistically significant at the 0.05 level.

如果两个模型的置信区间没有重叠，则表明两个模型的性能存在显著差异。

例如，假设我们有两个模型，它们的置信区间分别为 $[0.85, 0.95]$ 和 $[0.80, 0.90]$ 。由于这两个区间没有重叠，我们可以认为这两个模型的性能存在显著差异。

On the other hand, if two 95 percent confidence intervals overlap, we cannot automatically conclude that there's no significant difference between the two measurements. Even when confidence intervals overlap, there can still be a statistically significant difference.

如果两个模型的置信区间重叠，则不能自动得出两个测量值之间没有显著差异的结论。即使置信区间重叠，仍然可能存在统计显著差异。



Alternatively, to provide more detailed information about the exact quantities, we can use a table view to express the confidence intervals. The two common notations are summarized in Table [1.1](#).

Confidence Intervals

模型编号	置信区间 (\pm 表示法)	置信区间 (下限, 上限)
1	89.1% \pm 1.7%	89.1% (87.4%, 90.8%)
2	79.5% \pm 2.2%	79.5% (77.3%, 81.7%)
3	95.2% \pm 1.6%	95.2% (93.6%, 96.8%)

The \pm notation is often preferred if the confidence interval is *symmetric*, meaning the upper and lower endpoints are equidistant from the estimated parameter. Alternatively, the lower and upper confidence intervals can be written explicitly.

\pm 符号常用于表示置信区间，尤其是在置信区间是对称的情况下

The Methods

The following sections describe the four most common methods of constructing confidence intervals.

Method 1: Normal Approximation Intervals

The normal approximation interval involves generating the confidence interval from a single train-test split. It is often considered the simplest and most traditional method for computing confidence intervals. This approach is especially appealing in the realm of deep learning, where training models is computationally costly. It's also desirable when we are interested in evaluating a specific model, instead of models trained on various data partitions like in k -fold cross-validation.

How does it work? In short, the formula for calculating the confidence interval for a predicted parameter (for example, the sample mean, denoted as \bar{x}), assuming a



normal distribution, is expressed as $\bar{x} \pm z \times SE$.

In this formula, z represents the z -score, which indicates a particular value's number of standard deviations from the mean in a standard normal distribution. SE represents the standard error of the predicted parameter (in this case, the sample mean).

Most readers will be familiar with `z-score` tables that are usually found in the back of introductory statistics textbooks. However, a more convenient and preferred way to obtain `z-scores` is to use functions like SciPy's `stats.zscore` function, which computes the `z-scores` for given confidence levels.

For our scenario, the sample mean, denoted as \bar{x} , corresponds to the test set accuracy, ACC_{test} , a measure of successful predictions in the context of a binomial proportion confidence interval.

The standard error can be calculated under a normal approximation as follows:

$$SE = \sqrt{\frac{1}{n} ACC_{\text{test}} (1 - ACC_{\text{test}})}$$

In this equation, n signifies the size of the test set. Substituting the standard error back into the previous formula, we obtain the following:

$$ACC_{\text{test}} \pm z \sqrt{\frac{1}{n} ACC_{\text{test}} (1 - ACC_{\text{test}})}$$

Additional code examples to implement this method can also be found in the *supplementary/q25_confidence-intervals* subfolder in the supplementary code repository at <https://github.com/rasbt/MachineLearning-QandAI-book>. While the normal approximation interval method is very popular due to its simplicity, it has some downsides. First, the normal approximation may not always be accurate, especially for small sample sizes or for data that is not normally distributed. In such cases, other methods of computing confidence intervals may be more accurate. Second, using a single train-test split does not provide information about the variability of the model performance across different splits of the data. This can be an issue if the performance is highly dependent on the specific split used, which may be the case if the dataset is small or if there is a high degree of variability in the data.



Method 2: Bootstrapping Training Sets

Confidence intervals serve as a tool for approximating unknown parameters. However, when we are restricted to just one estimate, such as the accuracy derived from a single test set, we must make certain assumptions to make this work. For example, when we used the normal approximation interval described in the previous section, we assumed normally distributed data, which may or may not hold.

In a perfect scenario, we would have more insight into our test set sample distribution. However, this would require access to many independent test datasets, which is typically not feasible. A workaround is the bootstrap method, which resamples existing data to estimate the sampling distribution.

In practice, when the test set is large enough, the normal distribution approximation will hold, thanks to the central limit theorem. This theorem states that the sum (or average) of a large number of independent, identically distributed random variables will approach a normal distribution, regardless of the underlying distribution of the individual variables. It is difficult to specify what constitutes a large-enough test set. However, under stronger assumptions than those of the central limit theorem, we can at least estimate the rate of convergence to the normal distribution using the Berry-Esseen theorem, which gives a more quantitative estimate of how quickly the convergence in the central limit theorem occurs.

In a machine learning context, we can take the original dataset and draw a random sample *with replacement*. If the dataset has size n and we draw a random sample with replacement of size n , this implies that some data points will likely be duplicated in this new sample, whereas other data points are not sampled at all. We can then repeat this procedure for multiple rounds to obtain multiple training and test sets. This process is known as *out-of-bag bootstrapping*, illustrated in Figure [25.4](#).

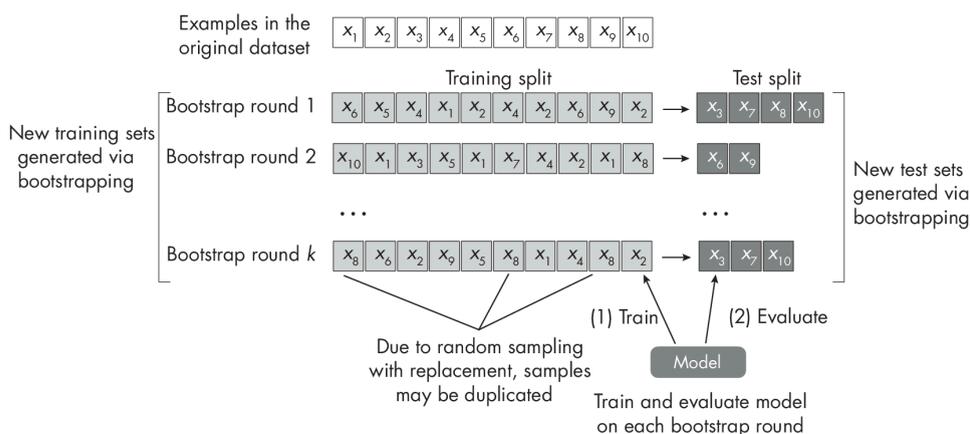


Figure 25.4

Suppose we constructed k training and test sets. We can now take each of these splits to train and evaluate the model to obtain k test set accuracy estimates. Considering this distribution of test set accuracy estimates, we can take the range between the 2.5th and 97.5th percentile to obtain the 95 percent confidence interval, as illustrated in Figure 25.5.

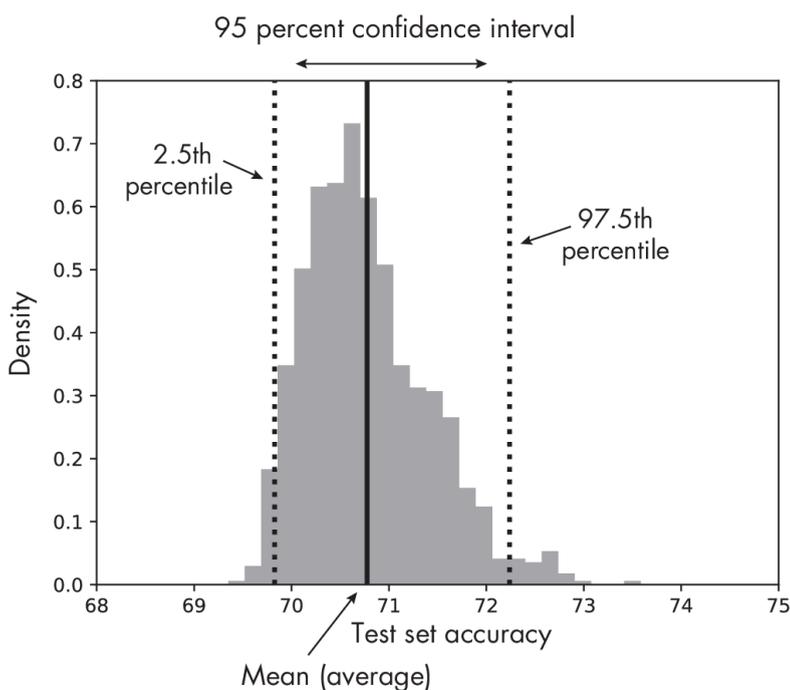


Figure 25.5

Unlike the normal approximation interval method, we can consider this out-of-bag bootstrap approach to be more agnostic to the specific distribution. Ideally, if the assumptions for the normal approximation are satisfied, both methodologies would yield identical outcomes.



Since bootstrapping relies on resampling the existing test data, its downside is that it doesn't bring in any new information that could be available in a broader population or unseen data. Therefore, it may not always be able to generalize the performance of the model to new, unseen data.

Note that we are using the bootstrap sampling approach in this chapter instead of obtaining the train-test splits via k -fold cross-validation, because of the bootstrap's theoretical grounding via the central limit theorem discussed earlier. There are also more advanced out-of-bag bootstrap methods, such as the .632 and .632+ estimates, which are reweighting the accuracy estimates.

Method 3: Bootstrapping Test Set Predictions

An alternative approach to bootstrapping training sets is to bootstrap test sets. The idea is to train the model on the existing training set as usual and then to evaluate the model on bootstrapped test sets, as illustrated in Figure 25.6. After obtaining the test set performance estimates, we can then apply the percentile method described in the previous section.

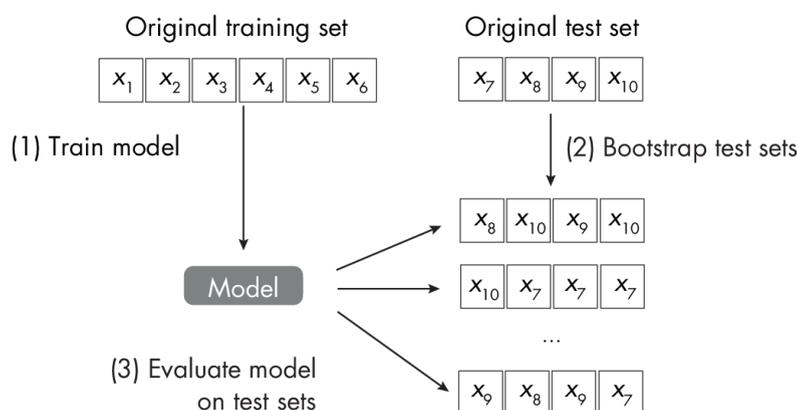


Figure 25.6

Contrary to the prior bootstrap technique, this method uses a trained model and simply resamples the test set (instead of the training sets). This approach is especially appealing for evaluating deep neural networks, as it doesn't require retraining the model on the new data splits. However, a disadvantage of this approach is that it doesn't assess the model's variability toward small changes in the training data.



Method 4: Retraining Models with Different Random Seeds

In deep learning, models are commonly retrained using various random seeds since some random weight initializations may lead to much better models than others. How can we build a confidence interval from these experiments? If we assume that the sample means follow a normal distribution, we can employ a previously discussed method where we calculate the confidence interval around a sample mean, denoted as \bar{x} , as follows:

$$\bar{x} \pm z \cdot \text{SE}$$

Since in this context we often work with a relatively modest number of samples (for instance, models from 5 to 10 random seeds), assuming a t distribution is deemed more suitable than a normal distribution. Therefore, we substitute the z value with a t value in the preceding formula. (As the sample size increases, the t distribution tends to look more like the standard normal distribution, and the critical values [z and t] become increasingly similar.)

Furthermore, if we are interested in the average accuracy, denoted as $\overline{\text{ACC}}_{\text{test}}$, we consider $\text{ACC}_{\text{test},j}$ corresponding to a unique random seed j as a sample. The number of random seeds we evaluate would then constitute the sample size n . As such, we would calculate:

$$\overline{\text{ACC}}_{\text{test}} \pm t \cdot \text{SE}$$

Here, SE is the standard error, calculated as $\text{SE} = \text{SD} / \sqrt{n}$, while

$$\overline{\text{ACC}}_{\text{test}} = \frac{1}{r} \sum_{j=1}^r \text{ACC}_{\text{test},j}$$

is the average accuracy, which we compute over the r random seeds. The standard deviation SD is calculated as follows:

$$\text{SD} = \sqrt{\frac{\sum_{j=1}^r (\text{ACC}_{\text{test},j} - \overline{\text{ACC}}_{\text{test}})^2}{r - 1}}$$

To summarize, calculating the confidence intervals using various random seeds is another effective alternative. However, it is primarily beneficial for deep learning models. It proves to be costlier than both the normal approximation approach

(method 1) and bootstrapping the test set (method 3), as it necessitates retraining the model. On the bright side, the outcomes derived from disparate random seeds provide us with a robust understanding of the model's stability.



Recommendations

Each possible method for constructing confidence intervals has its unique advantages and disadvantages. The normal approximation interval is cheap to compute but relies on the normality assumption about the distribution. The out-of-bag bootstrap is agnostic to these assumptions but is substantially more expensive to compute. A cheaper alternative is bootstrapping the test only, but this involves bootstrapping a smaller dataset and may be misleading for small or nonrepresentative test set sizes. Lastly, constructing confidence intervals from different random seeds is expensive but can give us additional insights into the model's stability.

Exercises

25-1. As mentioned earlier, the most common choice of confidence level is 95 percent confidence intervals. However, 90 percent and 99 percent are also common. Are 90 percent confidence intervals smaller or wider than 95 percent confidence intervals, and why is this the case?

25-2. In "???" on page , we created test sets by bootstrapping and then applied the already trained model to compute the test set accuracy on each of these datasets. Can you think of a method or modification to obtain these test accuracies more efficiently?

References

- A detailed discussion of the pitfalls of concluding statistical significance from nonoverlapping confidence intervals: Martin Krzywinski and Naomi Altman, "Error Bars"? (2013), <https://www.nature.com/articles/nmeth.2659>.
- A more detailed explanation of the binomial proportion confidence interval: https://en.wikipedia.org/wiki/Binomial_proportion_confidence_interval.



- For a detailed explanation of normal approximation intervals, see Section 1.7 of my article: "Model Evaluation, Model Selection, and Algorithm Selection in Machine Learning"? (2018), <https://arxiv.org/abs/1811.12808>.
- Additional information on the central limit theorem for independent and identically distributed random variables:
https://en.wikipedia.org/wiki/Central_limit_theorem.
- For more on the Berry-Esseen theorem:
https://en.wikipedia.org/wiki/Berry-Esseen_theorem.
- The .632 bootstrap addresses a pessimistic bias of the regular out-of-bag bootstrapping approach: Bradley Efron, "Estimating the Error Rate of a Prediction Rule: Improvement on Cross-Validation"? (1983),
<https://www.jstor.org/stable/2288636>.
- The .632+ bootstrap corrects an optimistic bias introduced in the .632 bootstrap: Bradley Efron and Robert Tibshirani, "Improvements on Cross-Validation: The .632+ Bootstrap Method"? (1997),
<https://www.jstor.org/stable/2965703>.
- A deep learning research paper that discusses bootstrapping the test set predictions: Benjamin Sanchez-Lengeling et al., "Machine Learning for Scent: Learning Generalizable Perceptual Representations of Small Molecules"? (2019), <https://arxiv.org/abs/1910.10685>.



Chapter 26: Confidence Intervals vs. Conformal Predictions

{#chapter-26-confidence-intervals-vs-conformal-predictions}

What are the differences between confidence intervals and conformal predictions, and when do we use one over the other?

Confidence intervals and conformal predictions are both statistical methods to estimate the range of plausible values for an unknown population parameter. As discussed in Chapter [\[ch25\]](#), a confidence interval quantifies the level of confidence that a population parameter lies within an interval. For instance, a 95 percent confidence interval for the mean of a population means that if we were to take many samples from the population and calculate the 95 percent confidence interval for each sample, we would expect the true population mean (average) to lie within these intervals 95 percent of the time. Chapter [\[ch25\]](#) covered several techniques for applying this method to estimate the prediction performance of machine learning models. Conformal predictions, on the other hand, are commonly used for creating prediction intervals, which are designed to cover a true outcome with a certain probability.

This chapter briefly explains what a prediction interval is and how it differs from confidence intervals, and then it explains how conformal predictions are, loosely speaking, a method for constructing prediction intervals.

Confidence Intervals and Prediction Intervals

Whereas a confidence interval focuses on parameters that characterize a population as a whole, a *prediction interval* provides a range of values for a single predicted target value. For example, consider the problem of predicting people's heights. Given a sample of 10,000 people from the population, we might conclude that the mean (average) height is 5 feet, 7 inches. We might also calculate a 95 percent confidence interval for this mean, ranging from 5 feet, 6 inches to 5 feet, 8 inches.



A *prediction interval*, however, is concerned with estimating not the height of the population but the height of an individual person. For example, given a weight of 185 pounds, a given person's prediction interval may fall between 5 feet 8 inches and 6 feet.

In a machine learning model context, we can use confidence intervals to estimate a population parameter such as the accuracy of a model (which refers to the performance on all possible prediction scenarios). In contrast, a prediction interval estimates the range of output values for a single given input example.

Prediction Intervals and Conformal Predictions

Both conformal predictions and prediction intervals are statistical techniques that estimate uncertainty for individual model predictions, but they do so in different ways and under different assumptions.

While prediction intervals often assume a particular data distribution and are tied to a specific type of model, conformal prediction methods are distribution free and can be applied to any machine learning algorithm.

In short, we can think of conformal predictions as a more flexible and generalizable form of prediction intervals. However, conformal predictions often require more computational resources than traditional methods for constructing prediction intervals, which involve resampling or permutation techniques.

Prediction Regions, Intervals, and Sets

In the context of conformal prediction, the terms *prediction interval*, *prediction set*, and *prediction region* are used to denote the plausible outputs for a given instance. The type of term used depends on the nature of the task.

In regression tasks where the output is a continuous variable, a *prediction interval* provides a range within which the true value is expected to fall with a certain level of confidence. For example, a model might predict that the price of a house is between \$200,000 and \$250,000.



In classification tasks, where the output is a discrete variable (the class labels), a *prediction set* includes all class labels that are considered plausible predictions for a given instance. For example, a model might predict that an image depicts either a cat, dog, or bird.

Prediction region is a more general term that can refer to either a prediction interval or a prediction set. It describes the set of outputs considered plausible by the model.

Computing Conformal Predictions

Now that we've introduced the difference between confidence intervals and prediction regions and learned how conformal prediction methods are related to prediction intervals, how exactly do conformal predictions work?

In short, conformal prediction methods provide a framework for creating prediction regions, sets of potential outcomes for a prediction task. Given the assumptions and methods used to construct them, these regions are designed to contain the true outcome with a certain probability.

For classifiers, a prediction region for a given input is a set of labels such that the set contains the true label with a given confidence (typically 95 percent), as illustrated in Figure 26.1.

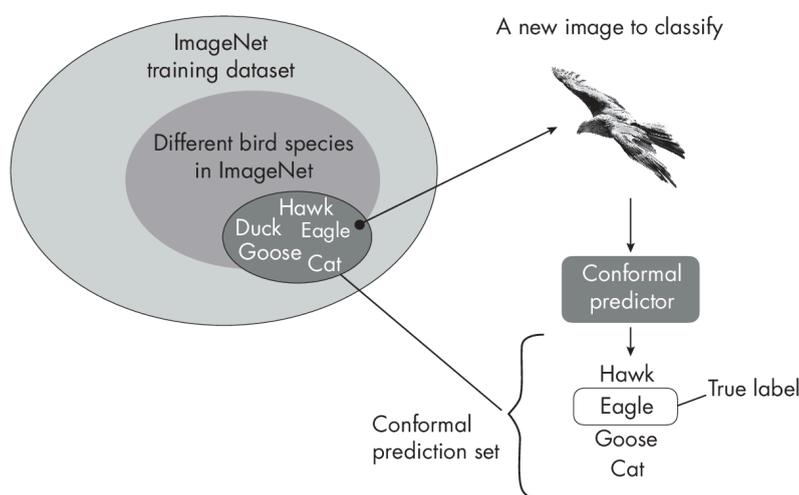


Figure 26.1

As depicted in Figure 26.1, the ImageNet dataset consists of a subset of bird species. Some bird species in ImageNet belong to one of the following classes:



hawk, duck, eagle, or goose. ImageNet also contains other animals, for example, cats. For a new image to classify (here, an eagle), the conformal prediction set consists of classes such that the true label, *eagle*, is contained within this set with 95 percent probability. Often, this includes closely related classes, such as *hawk* and *goose* in this case. However, the prediction set can also include less closely related class labels, such as *cat*.

To sketch the concept of computing prediction regions step by step, let's suppose we train a machine learning classifier for images. Before the model is trained, the dataset is typically split into three parts: a training set, a calibration set, and a test set. We use the training set to train the model and the calibration set to obtain the parameters for the conformal prediction regions. We can then use the test set to assess the performance of the conformal predictor. A typical split ratio might be 60 percent training data, 20 percent calibration data, and 20 percent test data.

The first step after training the model on the training set is to define a *nonconformity measure*, a function that assigns a numeric score to each instance in the calibration set based on how "unusual" it is. This could be based on the distance to the classifier's decision boundary or, more commonly, 1 minus the predicted probability of a class label. The higher the score is, the more unusual the instance is.

Before using conformal predictions for new data points, we use the nonconformity scores from the calibration set to compute a quantile threshold. This threshold is a probability level such that, for example, 95 percent of the instances in the calibration set (if we choose a 95 percent confidence level) have nonconformity scores below this threshold. This threshold is then used to determine the prediction regions for new instances, ensuring that the predictions are calibrated to the desired confidence level.

Once we have the threshold value, we can compute prediction regions for new data. Here, for each possible class label (each possible output of your classifier) for a given instance, we check whether its nonconformity score is below the threshold. If it is, then we include it in the prediction set for that instance.

A Conformal Prediction Example



Let's illustrate this process of making conformal predictions with an example using a simple conformal prediction method known as the *score method*. Suppose we train a classifier on a training set to distinguish between three species of birds: sparrows, robins, and hawks. Suppose the predicted probabilities for a calibration dataset are as follows:

- Sparrow [0.95, 0.9, 0.85, 0.8, 0.75]
- Robin [0.7, 0.65, 0.6, 0.55, 0.5]
- Hawk [0.4, 0.35, 0.3, 0.25, 0.2]

As depicted here, we have a calibration set consisting of 15 examples, five for each of the three classes. Note that a classifier returns three probability scores for each training example: one probability corresponding to each of the three classes (*Sparrow*, *Robin*, and *Hawk*). Here, however, we've selected only the probability for the true class label. For example, we may obtain the values [0.95, 0.02, 0.03] for the first calibration example with the true label *Sparrow*. In this case, we kept only 0.95.

Next, after we obtain the previous probability scores, we can compute the nonconformity score as 1 minus the probability, as follows:

- Sparrow [0.05, 0.1, 0.15, 0.2, 0.25]
- Robin [0.3, 0.35, 0.4, 0.45, 0.5]
- Hawk [0.6, 0.65, 0.7, 0.75, 0.8]

Considering a confidence level of 0.95, we now select a threshold such that 95 percent of these nonconformity scores fall below that threshold. Based on the nonconformity scores in this example, this threshold is 0.8. We can then use this threshold to construct the prediction sets for new instances we want to classify.

Now suppose we have a new instance (a new image of a bird) that we want to classify. We calculate the nonconformity score of this new bird image, assuming it belongs to each bird species (class label) in the training set:

- Sparrow 0.26
- Robin 0.45
- Hawk 0.9

In this case, the *Sparrow* and *Robin* nonconformity scores fall below the threshold of 0.8. Thus, the prediction set for this input is [*Sparrow*, *Robin*]. In other words,



this tells us that, on average, the true class label is included in the prediction set 95 percent of the time.

A hands-on code example implementing the score method can be found in the *supplementary/q26_conformal-prediction* subfolder at <https://github.com/rasbt/MachineLearning-QandAI-book>.

The Benefits of Conformal Predictions

In contrast to using class-membership probabilities returned from classifiers, the major benefits of conformal prediction are its theoretical guarantees and its generality. Conformal prediction methods don't make any strong assumptions about the distribution of the data or the model being used, and they can be applied in conjunction with any existing machine learning algorithm to provide confidence measures for predictions.

Confidence intervals have asymptotic coverage guarantees, which means that the coverage guarantee holds in the limit as the sample (test set) size goes to infinity. This doesn't necessarily mean that confidence intervals work for only very large sample sizes, but rather that their properties are more firmly guaranteed as the sample size increases. Confidence intervals therefore rely on asymptotic properties, meaning that their guarantees become more robust as the sample size grows.

In contrast, conformal predictions provide finite-sample guarantees, ensuring that the coverage probability is achieved for any sample size. For example, if we specify a 95 percent confidence level for a conformal prediction method and generate 100 calibration sets with corresponding prediction sets, the method will include the true class label for 95 out of the 100 test points. This holds regardless of the size of the calibration sets.

While conformal prediction has many advantages, it does not always provide the tightest possible prediction intervals. Sometimes, if the underlying assumptions of a specific classifier hold, that classifier's own probability estimates might offer tighter and more informative intervals.

Recommendations



A confidence interval tells us about our level of uncertainty about the model's properties, such as the prediction accuracy of a classifier. A prediction interval or conformal prediction output tells us about the level of uncertainty in a specific prediction from the model. Both are very important in understanding the reliability and performance of our model, but they provide different types of information.

For example, a confidence interval for the prediction accuracy of a model can be helpful for comparing and evaluating models and for deciding which model to deploy. On the other hand, a prediction interval can be helpful for using a model in practice and understanding its predictions. For instance, it can help identify cases where the model is unsure and may need additional data, human oversight, or a different approach.

Exercises

26-1. Prediction set sizes can vary between instances. For example, we may encounter a prediction set size of 1 for a given instance and for another, a set size of 3. What does the prediction set size tell us?

26-2. Chapters [ch25] and [ch26] focused on classification methods. Could we use conformal prediction and confidence intervals for regression too?

References

- MAPIE is a popular library for conformal predictions in Python: <https://mapie.readthedocs.io/>.
- For more on the score method used in this chapter: Christoph Molnar, *Introduction to Conformal Prediction with Python* (2023), <https://christophmolnar.com/books/conformal-prediction/>.
- In addition to the score method, several other variants of conformal prediction methods exist. For a comprehensive collection of conformal prediction literature and resources, see the Awesome Conformal Prediction page: <https://github.com/valeman/awesome-conformal-prediction>.





Chapter 27: Proper Metrics

What are the three properties of a distance function that make it a *proper metric*?

Metrics are foundational to mathematics, computer science, and various other scientific domains. Understanding the fundamental properties that define a good distance function to measure distances or differences between points or datasets is important. For instance, when dealing with functions like loss functions in neural networks, understanding whether they behave like proper metrics can be instrumental in knowing how optimization algorithms will converge to a solution.

Tips:

- **Metrics** 度量，是 **数学**、**计算机科学** 和各种其他科学领域的基础。
- 理解定义良好的 **距离函数** 的关键属性，对于测量点或数据集之间的距离或差异至关重要。
- 例如，在处理神经网络中的损失函数时，了解它们是否表现出良好的距离函数属性，对于了解优化算法如何收敛到解决方案至关重要。
- **距离函数**，具有三个关键属性：**非负性**、**对称性** 和 **三角不等式**。

This chapter analyzes two commonly utilized loss functions, the **mean squared error** and the **cross-entropy loss**, to demonstrate whether they meet the criteria for proper metrics.

Tips: 本章节，分析了两个常用的损失函数，**均方误差** 和 **交叉熵损失**，来演示它们是否符合良好的度量标准。

The Criteria

To illustrate the criteria of a proper metric, consider two vectors or points \mathbf{v} and \mathbf{w} , and their distance $d(\mathbf{v}, \mathbf{w})$, as shown in Figure [27.1](#).

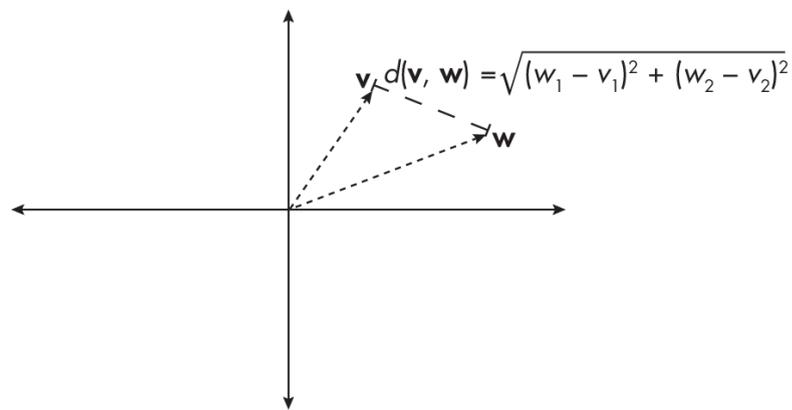


Figure 27.1

The criteria of a proper metric are the following:

- The distance between two points is always **non-negative**, $d(\mathbf{v}, \mathbf{w}) \geq 0$, and can be 0 only if the two points are identical, that is, $\mathbf{v} = \mathbf{w}$.
- The distance is **symmetric**; for instance, $d(\mathbf{v}, \mathbf{w}) = d(\mathbf{w}, \mathbf{v})$.
- The distance function satisfies the **triangle inequality** for any three points: \mathbf{v} , \mathbf{w} , \mathbf{x} , meaning:

$$d(\mathbf{v}, \mathbf{w}) \leq d(\mathbf{v}, \mathbf{x}) + d(\mathbf{x}, \mathbf{w})$$

Tips: 距离函数，具有三个关键属性：**非负性**、**对称性** 和 **三角不等式**。

To better understand the triangle inequality, think of the points as vertices of a triangle. If we consider any triangle, the sum of two of the sides is always larger than the third side, as illustrated in Figure [27.2](#).

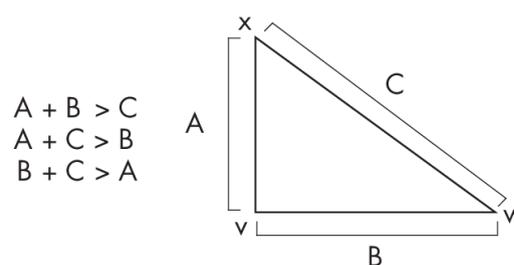


Figure 27.2

Consider what would happen if the triangle inequality depicted in Figure [27.2](#) weren't true. If the sum of the lengths of sides AB and BC was shorter than AC, then sides AB and BC would not meet to form a triangle; instead, they would fall



short of each other. Thus, the fact that they meet and form a triangle demonstrates the triangle inequality.

The Mean Squared Error

The **mean squared error (MSE)** loss computes the squared Euclidean distance between a target variable y and a predicted target value \hat{y} :

$$\text{MSE} = \frac{1}{n} \sum_{i=1}^n \left(y^{(i)} - \hat{y}^{(i)} \right)^2$$

The index i denotes the i th data point in the dataset or sample. Is this loss function a proper metric?

For simplicity's sake, we will consider the **squared error (SE)** loss between two data points (though the following insights also hold for the MSE). As shown in the following equation, the SE loss quantifies the squared difference between the predicted and actual values for a single data point, while the MSE loss averages these squared differences over all data points in a dataset:

$$\text{SE}(y, \hat{y}) = (y - \hat{y})^2$$

In this case, the SE satisfies the first part of the first criterion: the distance between two points is always **non-negative**. Since we are raising the difference to the power of 2, it cannot be negative.

How about the second criterion, that the distance can be 0 only if the two points are identical? Due to the subtraction in the SE, it is intuitive to see that it can be 0 only if the prediction matches the target variable, $y = \hat{y}$. As with the first criterion, we can use the square to confirm that SE satisfies the second criterion: we have $(y - \hat{y})^2 = (\hat{y} - y)^2$.

At first glance, it seems that the squared error loss also satisfies the third criterion, the **triangle inequality**. Intuitively, you can check this by choosing three arbitrary numbers, here 1, 2, 3:

- $(1 - 2)^2 \leq (1 - 3)^2 + (2 - 3)^2$
- $(1 - 3)^2 \leq (1 - 2)^2 + (2 - 3)^2$



$$\bullet (2 - 3)^2 \leq (1 - 2)^2 + (1 - 3)^2$$

However, there are values for which this is not true. For example, consider the values $a = 0$, $b = 2$, and $c = 1$. This gives us $d(a, b) = 4$, $d(a, c) = 1$, and $d(b, c) = 1$, such that we have the following scenario, which violates the triangle inequality:

$$\bullet (0 - 2)^2 \not\leq (0 - 1)^2 + (2 - 1)^2$$

$$\bullet (2 - 1)^2 \leq (0 - 1)^2 + (0 - 2)^2$$

$$\bullet (0 - 1)^2 \leq (0 - 2)^2 + (1 - 2)^2$$

Since it does not satisfy the triangle inequality via the example above, we conclude that the (mean) squared error loss is not a proper metric.

However, if we change the squared error into the **root-squared error**

$$\sqrt{(y - \hat{y})^2}$$

the triangle inequality can be satisfied:

$$\sqrt{(0 - 2)^2} \leq \sqrt{(0 - 1)^2} + \sqrt{(2 - 1)^2}$$

You might be familiar with the **L2** distance or Euclidean distance, which is known to satisfy the triangle inequality. These two distance metrics are equivalent to the root-squared error when considering two scalar values.

Tips:

- 如果将 **平方误差** 改为 **平方根误差**，则三角不等式可以满足。
- 平方根误差，是平方误差的平方根。

The Cross-Entropy Loss

Tips: 交叉熵损失，是衡量两个概率分布之间距离的损失函数。

FIXME ??? 不理解



Cross entropy is used to measure the distance between two probability distributions. In machine learning contexts, we use the discrete cross-entropy loss (CE) between class label y and the predicted probability \hat{p} when we train logistic regression or neural network classifiers on a dataset consisting of n training examples:

$$\text{CE}(\mathbf{y}, \mathbf{p}) = -\frac{1}{n} \sum_{i=1}^n y^{(i)} \times \log(p^{(i)})$$

Is this loss function a proper metric? Again, for simplicity's sake, we will look at the cross-entropy function (H) between only two data points:

$$H(y, p) = -y \times \log(p)$$

The cross-entropy loss satisfies one part of the first criterion: the distance is always non-negative because the probability score is a number in the range $[0, 1]$. Hence, $\log(p)$ ranges between $-\infty$ and 0. The important part is that the H function includes a negative sign. Hence, the cross entropy ranges between 0 and $+\infty$ and thus satisfies one aspect of the first criterion shown above.

However, the cross-entropy loss is not 0 for two identical points. For example, $H(0.9, 0.9) = -0.9 \times \log(0.9) = 0.095$.

The second criterion shown above is also violated by the cross-entropy loss because the loss is not symmetric: $-y \times \log(p) \neq -p \times \log(y)$. Let's illustrate this with a concrete, numeric example:

- If $y = 1$ and $p = 0.5$, then $-1 \times \log(0.5) = 0.693$.
- If $y = 0.5$ and $p = 1$, then $-0.5 \times \log(1) = 0$.

Finally, the cross-entropy loss does not satisfy the triangle inequality, $H(r, p) \geq H(r, q) + H(q, p)$. Let's illustrate this with an example as well. Suppose we choose $r = 0.9$, $p = 0.5$, and $q = 0.4$. We have:

- $H(0.9, 0.5) = 0.624$
- $H(0.9, 0.4) = 0.825$
- $H(0.4, 0.5) = 0.277$

As you can see, $0.624 \geq 0.825 + 0.277$ does not hold here.



In conclusion, while the **cross-entropy loss** is a useful loss function for training neural networks via (stochastic) gradient descent, it is not a proper distance metric, as it does not satisfy any of the three criteria.

Tips:

- 交叉熵损失，是训练逻辑回归或神经网络分类器时，用于衡量两个概率分布之间距离的损失函数，这种损失函数在训练过程中，可以引导模型学习到更好的概率分布。
- 但是，交叉熵损失，不是良好的度量标准，因为它不满足三角不等式。

Exercises

27-1. Suppose we consider using the mean absolute error (MAE) as an alternative to the root mean square error (RMSE) for measuring the performance of a machine learning model, where

$$\text{MAE} = \frac{1}{n} \sum_{i=1}^n |y^{(i)} - \hat{y}^{(i)}|$$

and

$$\text{RMSE} = \sqrt{\frac{1}{n} \sum_{i=1}^n (y^{(i)} - \hat{y}^{(i)})^2}$$

However, a colleague argues that the MAE is not a proper distance metric in metric space because it involves an absolute value, so we should use the RMSE instead. Is this argument correct?

27-2. Based on your answer to the previous question, would you say that the MAE is better or is worse than the RMSE?



Chapter 28: The k in k -Fold Cross-Validation

k -fold cross-validation is a common choice for evaluating machine learning classifiers because it lets us use all training data to simulate how well a machine learning algorithm might perform on new data. What are the advantages and disadvantages of choosing a large k ?

本章讨论了 k -fold 交叉验证，并讨论了它的优缺点。

- k -fold 交叉验证是一种常用的评估机器学习分类器的方法，它让我们使用所有训练数据来模拟机器学习算法在新数据上的表现。
- 选择较大的 k 值时，训练集之间的差异较小，因此模型之间的差异也较小。
- 选择较小的 k 值时，训练集之间的差异较大，因此模型之间的差异也较大。

We can think of k -fold cross-validation as a workaround for model evaluation when we have limited data. In machine learning model evaluation, we care about the generalization performance of our model, that is, how well it performs on new data. In k -fold cross-validation, we use the training data for model selection and evaluation by partitioning it into k validation rounds and folds. If we have k folds, we have k iterations, leading to k different models, as illustrated in Figure 28.1.

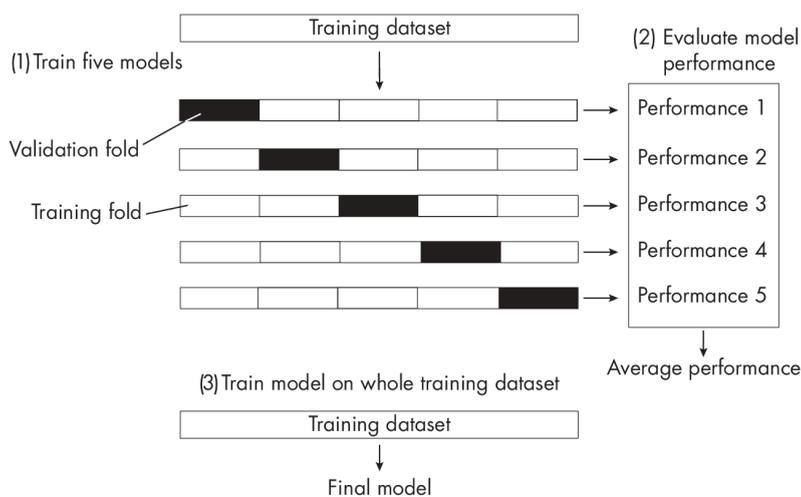


Figure 28.1



Using k -fold cross-validation, we usually evaluate the performance of a particular hyperparameter configuration by computing the average performance over the k models. This performance reflects or approximates the performance of a model trained on the complete training dataset after evaluation.

使用 k -fold 交叉验证，我们通常通过计算 k 个模型的平均性能，来评估特定超参配置的性能。

The following sections cover the trade-offs of selecting values for k in k -fold cross-validation and address the challenges of large k values and their computational demands, especially in deep learning contexts. We then discuss the core purposes of k and how to choose an appropriate value based on specific modeling needs.

Trade-offs in Selecting Values for k

If k is too large, the training sets are too similar between the different rounds of cross-validation. The k models are thus very similar to the model we obtain by training on the whole training set. In this case, we can still leverage the advantage of k -fold cross-validation: evaluating the performance for the entire training set via the held-out validation fold in each round. (Here, we obtain the training set by concatenating all $k - 1$ training folds in a given iteration.) However, a disadvantage of a large k is that it is more challenging to analyze how the machine learning algorithm with the particular choice of hyperparameter setting behaves on different training datasets.

Besides the issue of too-similar datasets, running k -fold cross-validation with a large value of k is also computationally more demanding. A larger k is more expensive since it increases both the number of iterations and the training set size at each iteration. This is especially problematic if we work with relatively large models that are expensive to train, such as contemporary deep neural networks.

A common choice for k is typically 5 or 10, for practical and historical reasons. A study by Ron Kohavi (see Reference at the end of this chapter) found that $k = 10$ offers a good bias and variance trade-off for classical machine learning algorithms, such as decision trees and naive Bayes classifiers, on a handful of small datasets.

5 或 10 是 k -fold 交叉验证的常见选择，这是出于实际和历史原因。Ron Kohavi 的研究（见本章末尾的参考文献）发现， $k = 10$ 在小型数据集上对经典机器学习算法（如决策树和朴素贝叶斯分类器）提供了良好的偏差和方差权衡。

For example, in 10-fold cross-validation, we use 9/10 (90 percent) of the data for training in each round, whereas in 5-fold cross-validation, we use only 4/5 (80 percent) of the data, as shown in Figure 28.2.

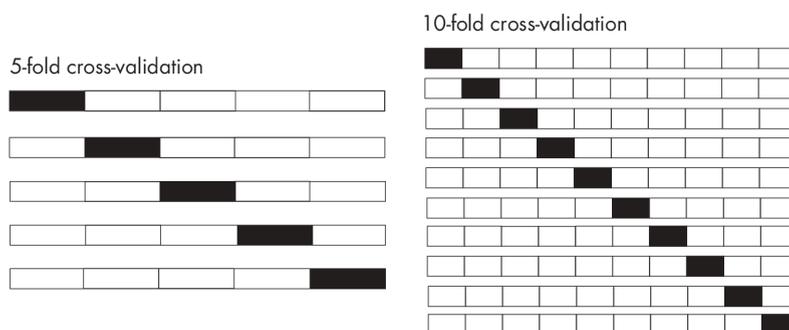


Figure 28.2

However, this does not mean large training sets are bad, since they can reduce the pessimistic bias of the performance estimate (mostly a good thing) if we assume that the model training can benefit from more training data. (See Figure 5.1 on page for an example of a learning curve.)

In practice, both a very small and a very large k may increase variance. For instance, a larger k makes the training folds more similar to each other since a smaller proportion is left for the held-out validation sets. Since the training folds are more similar, the models in each round will be more similar. In practice, we may observe that the variance of the held-out validation fold scores is more similar for larger values of k . On the other hand, when k is large, the validation sets are small, so they may contain more random noise or be more susceptible to quirks of the data, leading to more variation in the validation scores across the different folds. Even though the models themselves are more similar (since the training sets are more similar), the validation scores may be more sensitive to the particularities of the small validation sets, leading to higher variance in the overall cross-validation score.

k 过大或过小，都会增加方差。



Determining Appropriate Values for k

When deciding upon an appropriate value of k , we are often guided by computational performance and conventions. However, it's worthwhile to define the purpose and context of using k -fold cross-validation. For example, if we care primarily about approximating the predictive performance of the final model, using a large k makes sense. This way, the training folds are very similar to the combined training dataset, yet we still get to evaluate the model on all data points via the validation folds.

决定适当的 k 值时，先考虑使用 k -fold 交叉验证的目的和上下文。例如，如果我们主要关心近似最终模型的预测性能，使用较大的 k 是有意义的。这样，训练集非常类似于组合训练数据集，但我们仍然可以通过验证集评估模型。

On the other hand, if we care to evaluate how sensitive a given hyperparameter configuration and training pipeline is to different training datasets, then choosing a smaller number for k makes more sense.

如果我们主要关心给定超参配置和训练管道对不同训练数据集的敏感性，那么选择较小的 k 是有意义的。

Since most practical scenarios consist of two steps -- tuning hyperparameters and evaluating the performance of a model -- we can also consider a two-step procedure. For instance, we can use a smaller k during hyperparameter tuning. This will help speed up the hyperparameter search and probe the hyperparameter configurations for robustness (in addition to the average performance, we can also consider the variance as a selection criterion). Then, after hyperparameter tuning and selection, we can increase the value of k to evaluate the model.

大多数实际场景都包括两个步骤：调整超参和评估模型性能。因此，我们也可以考虑一个两步流程。

- 例如，在调整超参时，我们可以使用较小的 k 。这将帮助加速超参搜索，并测试超参配置的稳健性（除了平均性能，我们还可以考虑方差作为选择标准）。
- 然后，在调整超参和选择后，我们可以增加 k 值来评估模型。



However, reusing the same dataset for model selection and evaluation introduces biases, and it is usually better to use a separate test set for model evaluation. Also, **nested cross-validation** may be preferred as an alternative to k -fold cross-validation.

然而，重复使用相同的数据集进行模型选择和评估，会引入偏差，通常最好使用单独的测试集进行模型评估。此外，嵌套交叉验证可能比 k -fold 交叉验证更可取。

更多细节：[嵌套交叉验证](#)

Exercises

28-1. Suppose we want to provide a model with as much training data as possible. We consider using *leave-one-out cross-validation (LOOCV)*, a special case of k -fold cross-validation where k is equal to the number of training examples, such that the validation folds contain only a single data point. A colleague mentions that LOOCV is defective for discontinuous loss functions and performance measures such as classification accuracy. For instance, for a validation fold consisting of only one example, the accuracy is always either 0 (0 percent) or 1 (99 percent). Is this really a problem?

28-2. This chapter discussed model selection and model evaluation as two use cases of k -fold cross-validation. Can you think of other use cases?

References

- For a longer and more detailed explanation of why and how to use k -fold cross-validation, see my article: "Model Evaluation, Model Selection, and Algorithm Selection in Machine Learning"? (2018), <https://arxiv.org/abs/1811.12808>.
- The paper that popularized the recommendation of choosing $k = 5$ and $k = 10$: Ron Kohavi, "A Study of Cross-Validation and Bootstrap for Accuracy Estimation and Model Selection"? (1995), <https://dl.acm.org/doi/10.5555/1643031.1643047>.





Chapter 29: Training and Test Set Discordance

Suppose we train a model that performs much better on the test dataset than on the training dataset. Since a similar model configuration previously worked well on a similar dataset, we suspect something might be unusual with the data. What are some approaches for looking into training and test set discrepancies, and what strategies can we use to mitigate these issues?

Before investigating the datasets in more detail, we should check for technical issues in the **data loading** and **evaluation** code. For instance, a simple sanity check is to temporarily replace the test set with the training set and to reevaluate the model. In this case, we should see identical training and test set performances (since these datasets are now identical). If we notice a discrepancy, we likely have a bug in the code; in my experience, such bugs are frequently related to incorrect shuffling or inconsistent (often missing) data normalization.

在进一步检查数据集之前，我们应该检查数据加载和评估代码中的技术问题。

- 例如，一个简单的健全性检查是暂时将测试集替换为训练集，并重新评估模型。
- 在这种情况下，我们应该看到训练和测试集的性能相同（因为这些数据集现在相同）。
- 如果我们注意到差异，我们代码可能有bug；根据经验，这种错误，通常是数据洗牌不均匀或数据归一化不一致（通常缺失）。

If the test set performance is much better than the training set performance, we can rule out overfitting. More likely, there are substantial differences in the training and test data distributions. These distributional differences may affect both the features and the targets. Here, plotting the target or label distributions of training and test data is a good idea. For example, a common issue is that the test set is missing certain class labels if the dataset was not shuffled properly before splitting it into training and test data. For small tabular datasets, it is also feasible to compare feature distributions in the training and test sets using histograms.



Looking at feature distributions is a good approach for tabular data, but this is trickier for image and text data. A relatively easy and more general approach to check for discrepancies between training and test sets is adversarial validation.

Adversarial validation, illustrated in

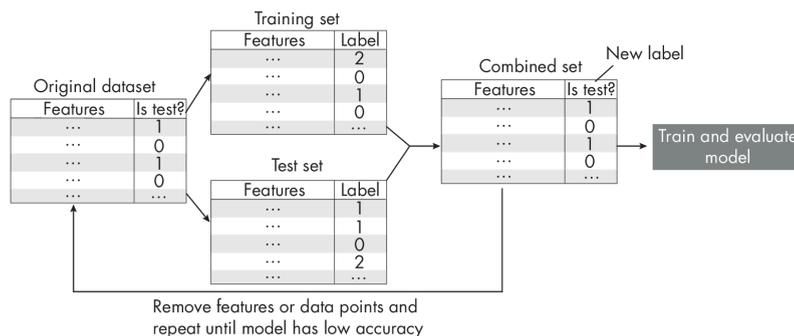


Figure 29.1

is a technique to identify the degree of similarity between the training and test data. We first merge the training and test sets into a single dataset, and then we create a binary target variable that distinguishes between training and test data. For instance, we can use a new *Is test?* label where we assign the label 0 to training data and the label 1 to test data. We then use *k*-fold cross-validation or repartition the dataset into a training set and a test set and train a machine learning model as usual. Ideally, we want the model to perform poorly, indicating that the training and test data distributions are similar. On the other hand, if the model performs well in predicting the *Is test?* label, it suggests a discrepancy between the training and test data that we need to investigate further.

对抗验证，用于识别训练和测试数据之间的相似程度。

- 我们首先将训练和测试集合并为一个数据集，然后创建一个二元目标变量，用于区分训练和测试数据。
- 例如，我们可以使用一个新的 *Is test?* 标签，将标签 0 分配给训练数据，将标签 1 分配给测试数据。
- 然后，我们使用 *k*-fold 交叉验证或重新划分数据集为训练集和测试集，并像往常一样训练机器学习模型。
- 理想情况下，我们希望模型表现不佳，表明训练和测试数据分布相似。
- 另一方面，如果模型在预测 *Is test?* 标签时表现良好，则表明训练和测试数据之间存在差异，我们需要进一步调查。



What mitigation techniques should we use if we detect a training-test set discrepancy using adversarial validation? If we're working with a tabular dataset, we can remove features one at a time to see if this helps address the issue, as spurious features can sometimes be highly correlated with the target variable. To implement this strategy, we can use sequential feature selection algorithms with an updated objective. For example, instead of maximizing classification accuracy, we can minimize classification accuracy. For cases where removing features is not so trivial (such as with image and text data), we can also investigate whether removing individual training instances that are different from the test set can address the discrepancy issue.

如果我们使用对抗验证检测到训练-测试集差异，我们应该使用什么缓解技术？如果我们使用表格数据集，我们可以一次删除一个特征，看看是否有助于解决这个问题，因为虚假特征有时与目标变量高度相关。为了实现这个策略，我们可以使用顺序特征选择算法，并更新目标函数。例如，我们不再最大化分类准确率，而是最小化分类准确率。对于图像和文本数据，我们也可以研究是否删除与测试集不同的训练实例是否有助于解决差异问题。

Exercises

29-1. What is a good performance baseline for the adversarial prediction task?

29-2. Since training datasets are often bigger than test datasets, adversarial validation often results in an imbalanced prediction problem (with a majority of examples labeled as *is test?* being false instead of true). Is this an issue, and if so, how can we mitigate that?



Chapter 30: Limited Labeled Data

Suppose we plot a learning curve (as shown in Figure 5.1 on page , for example) and find the machine learning model overfits and could benefit from more training data. What are some different approaches for dealing with limited labeled data in supervised machine learning settings?

学习曲线（Learning Curve）是机器学习中用于评估模型性能随训练数据量变化趋势的图表。它通常用于诊断模型是否存在过拟合或欠拟合问题。

In lieu of collecting more data, there are several methods related to regular supervised learning that we can use to improve model performance in limited labeled data regimes.

除了收集更多数据，还有几种方法，用于改进标签数据有限时的模型性能。

Improving Model Performance with Limited Labeled Data

The following sections explore various machine learning paradigms that help in scenarios where training data is limited.

Labeling More Data

Collecting additional training examples is often the best way to improve the performance of a model (a learning curve is a good diagnostic for this). However, this is often not feasible in practice, because acquiring high-quality data can be costly, computational resources and storage might be insufficient, or the data may be hard to access.

收集更多训练数据，通常是提高模型性能的最佳方法（学习曲线是诊断此问题的一个很好的指标）。

然而，这在实践中通常不可行，因为获取高质量数据可能很昂贵，计算资源和存储空间可能不足，或者数据可能难以获取。



Bootstrapping the Data

Similar to the techniques for reducing overfitting discussed in Chapter [\[ch05\]](#), it can be helpful to "bootstrap" the data by generating modified (augmented) or artificial (synthetic) training examples to boost the performance of the predictive model. Of course, improving the quality of data can also lead to the improved predictive performance of a model, as discussed in Chapter [\[ch21\]](#).

与第 5 章讨论的减少过拟合的技术类似，可以通过生成修改（增强）或人工（合成）训练示例来“引导”数据，以提高预测模型的性能。

当然，提高数据质量也可以提高模型的预测性能，如第 21 章所述。

Transfer Learning

Transfer learning describes training a model on a general dataset (for example, ImageNet) and then fine-tuning the pretrained target dataset (for example, a dataset consisting of different bird species), as outlined in Figure [30.1](#).

迁移学习描述了在通用数据集（例如 ImageNet）上训练模型，然后对预训练的目标数据集（例如包含不同鸟类物种的数据集）进行微调，如图 1.1 所示。

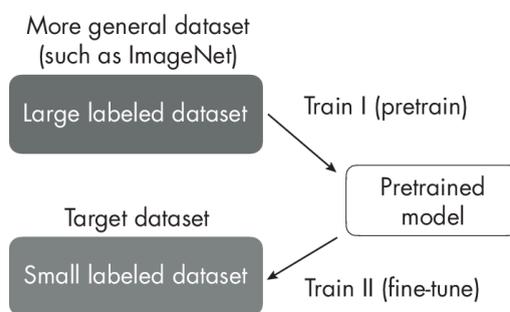


Figure 30.1

Transfer learning is usually done in the context of deep learning, where model weights can be updated. This is in contrast to tree-based methods, since most decision tree algorithms are nonparametric models that do not support iterative training or parameter updates.

迁移学习，通常用于深度学习场景，其中可以更新模型权重。这与基于树的方法形成对比，因为大多数决策树算法是非参数模型，不支持迭代训练或参数更新。



Self-Supervised Learning

Similar to transfer learning, in self-supervised learning, the model is pretrained on a different task before being fine-tuned to a target task for which only limited data exists. However, self-supervised learning usually relies on label information that can be directly and automatically extracted from unlabeled data. Hence, self-supervised learning is also often called **unsupervised pretraining**.

与迁移学习类似，在自监督学习中，模型在不同的任务上进行预训练，然后针对目标任务进行微调，而目标任务只有有限的数据。

然而，自监督学习通常依赖于可以直接从无标签数据中自动提取的标签信息。因此，自监督学习也经常被称为**无监督预训练**。

Common examples of self-supervised learning include the *next word* (used in GPT, for example) or *masked word* (used in BERT, for example) pretraining tasks in language modeling, covered in more detail in Chapter [\[ch17\]](#). Another intuitive example from computer vision includes *inpainting*: predicting the missing part of an image that was randomly removed, illustrated in Figure [30.2](#).

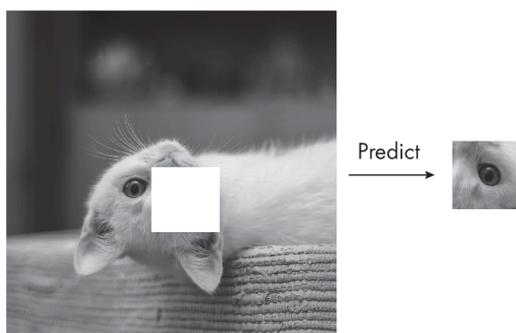


Figure 30.2

For more detail on self-supervised learning, see Chapter [\[ch02\]](#).

Active Learning



In active learning, illustrated in Figure 30.3, we typically involve manual labelers or users for feedback during the learning process. However, instead of labeling the entire dataset up front, active learning includes a prioritization scheme for suggesting unlabeled data points for labeling to maximize the machine learning model's performance.

在主动学习中，如图 1.3 所示，我们通常涉及手动标签器或用户在训练过程中提供反馈。

然而，与提前标记整个数据集不同，主动学习包括一个优先级方案，用于建议未标记的数据点进行标记，以最大化机器学习模型的性能。

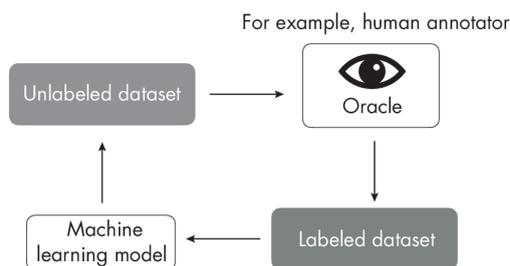


Figure 30.3

The term **active learning** refers to the fact that the model actively selects data for labeling. For example, the simplest form of active learning selects data points with high prediction uncertainty for labeling by a human annotator (also referred to as an *oracle*).

Few-Shot Learning

In a **few-shot** learning scenario, we often deal with extremely small datasets that include only a handful of examples per class. In research contexts, 1-shot (one example per class) and 5-shot (five examples per class) learning scenarios are very common. An extreme case of few-shot learning is **zero-shot** learning, where no labels are provided. Popular examples of zero-shot learning include GPT-3 and related language models, where the user has to provide all the necessary information via the input prompt, as illustrated in Figure 30.4.

在少样本学习场景中，我们通常处理包含每个类别只有少量示例的极端情况。

在研究上下文中，1-shot（每个类别一个示例）和5-shot（每个类别五个示例）学习场景非常常见。

小样本学习的极端情况，是零样本学习，其中没有提供标签。

零样本学习的流行示例包括 GPT-3 和相关语言模型，其中用户必须通过输入提示提供所有必要信息，如图 1.4 所示。



Zero-shot classification with ChatGPT

For more detail on few-shot learning, see Chapter [\[ch03\]](#).

Meta-Learning

Meta-learning involves developing methods that determine how machine learning algorithms can best learn from data. We can therefore think of meta-learning as "learning to learn." The machine learning community has developed several approaches for meta-learning. Within the machine learning community, the term *meta-learning* doesn't just represent multiple subcategories and approaches; it is also occasionally employed to describe related yet distinct processes, leading to nuances in its interpretation and application.

Meta-learning is one of the main subcategories of few-shot learning. Here, the focus is on learning a good feature extraction module, which converts support and query images into vector representations. These vector representations are optimized for determining the predicted class of the query example via comparisons with the training examples in the support set. (This form of meta-learning is illustrated in Chapter [\[ch03\]](#) on page .) Another branch of meta-learning unrelated to the few-shot learning approach is focused on extracting metadata (also called *meta-features*) from datasets for supervised learning tasks, as illustrated in Figure [30.5](#). The meta-features are descriptions of the dataset itself. For example, these can include the number of features and statistics of the different features (kurtosis, range, mean, and so on).

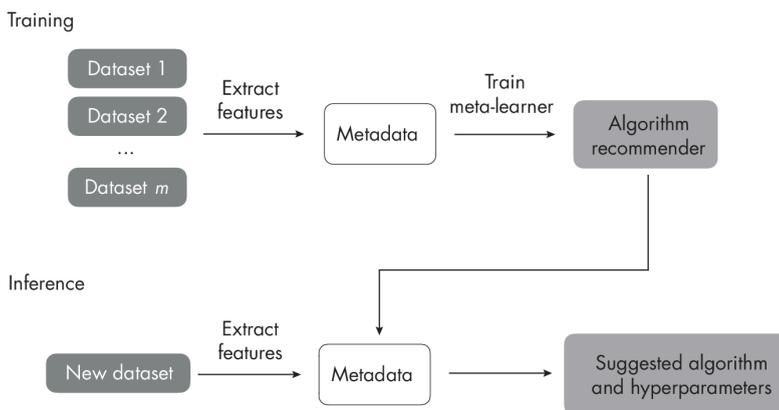


Figure 30.5

The extracted meta-features provide information for selecting a machine learning algorithm for the dataset at hand. Using this approach, we can narrow down the algorithm and hyperparameter search spaces, which helps reduce overfitting when the dataset is small.

Weakly Supervised Learning

Weakly supervised learning, illustrated in Figure 30.6, involves using an external label source to generate labels for an unlabeled dataset. Often, the labels created by a weakly supervised labeling function are more noisy or inaccurate than those produced by a human or domain expert, hence the term *weakly* supervised. We can develop or adopt a rule-based classifier to create the labels in weakly supervised learning; these rules usually cover only a subset of the unlabeled dataset.

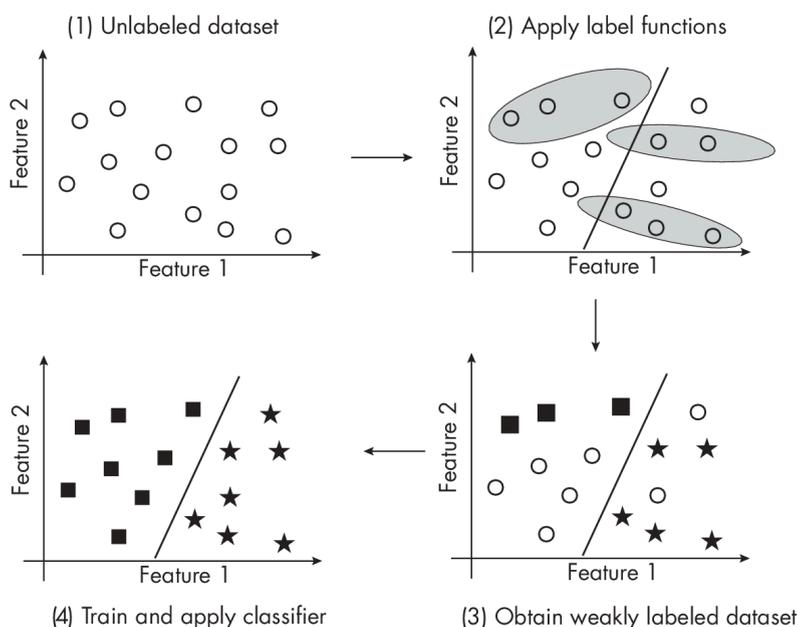


Figure 30.6



Let's return to the example of email spam classification from Chapter [ch23] to illustrate a rule-based approach for data labeling. In weak supervision, we could design a rule-based classifier based on the keyword *SALE* in the email subject header line to identify a subset of spam emails. Note that while we may use this rule to label certain emails as spam positive, we should not apply this rule to label emails without *SALE* as non-spam. Instead, we should either leave those unlabeled or apply a different rule to them.

There is a subcategory of weakly supervised learning referred to as PU-learning. In *PU-learning*, which is short for *positive-unlabeled learning*, we label and learn only from positive examples.

Semi-Supervised Learning

Semi-supervised learning is closely related to weakly supervised learning: it also involves creating labels for unlabeled instances in the dataset. The main difference between these two methods lies in *how* we create the labels. In weak supervision, we create labels using an external labeling function that is often noisy, inaccurate, or covers only a subset of the data. In semi-supervision, we do not use an external label function; instead, we leverage the structure of the data itself. We can, for example, label additional data points based on the density of neighboring labeled data points, as illustrated in Figure 30.7.

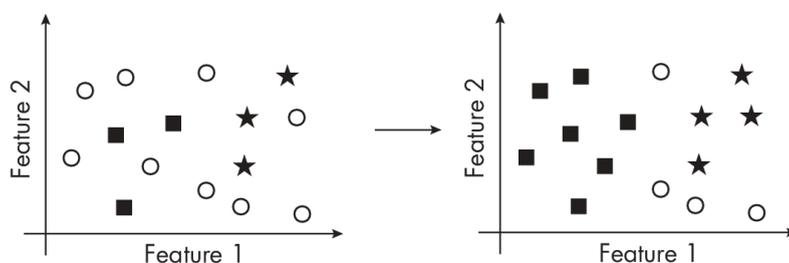


Figure 30.7

While we can apply weak supervision to an entirely unlabeled dataset, semi-supervised learning requires at least a portion of the data to be labeled. In practice, it is possible first to apply weak supervision to label a subset of the data and then to use semi-supervised learning to label instances that were not captured by the labeling functions.



Thanks to their close relationship, semi-supervised learning is sometimes referred to as a subcategory of weakly supervised learning, and vice versa.

Self-Training

Self-training falls somewhere between semi-supervised learning and weakly supervised learning. For this technique, we train a model to label the dataset or adopt an existing model to do the same. This model is also referred to as a *pseudo-labeler*.

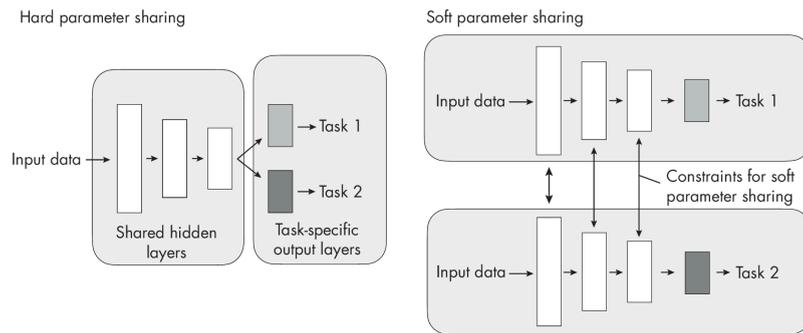
Self-training does not guarantee accurate labels and is thus related to weakly supervised learning. Moreover, while we use or adopt a machine learning model for this pseudo-labeling, self-training is also related to semi-supervised learning.

An example of self-training is knowledge distillation, discussed in Chapter [\[ch06\]](#).

Multi-Task Learning

Multi-task learning trains neural networks on multiple, ideally related tasks. For example, if we are training a classifier to detect spam emails, spam classification is the main task. In multi-task learning, we can add one or more related tasks for the model to solve, referred to as *auxiliary tasks*. For the spam email example, an auxiliary task could be classifying the email's topic or language.

Typically, multi-task learning is implemented via multiple loss functions that have to be optimized simultaneously, with one loss function for each task. The auxiliary tasks serve as an inductive bias, guiding the model to prioritize hypotheses that can explain multiple tasks. This approach often results in models that perform better on unseen data. There are two subcategories of multi-task learning: multi-task learning with hard parameter sharing and multi-task learning with soft parameter sharing. Figure [30.8](#) illustrates the difference between these two methods.

**Figure 30.8**

In *hard* parameter sharing, as shown in Figure 30.8, only the output layers are task specific, while all the tasks share the same hidden layers and neural network backbone architecture. In contrast, *soft* parameter sharing uses separate neural networks for each task, but regularization techniques such as distance minimization between parameter layers are applied to encourage similarity among the networks.

Multimodal Learning

While multi-task learning involves training a model with multiple tasks and loss functions, multimodal learning focuses on incorporating multiple types of input data.

Common examples of multimodal learning are architectures that take both image and text data as input (though multimodal learning is not restricted to only two modalities and can be used for any number of input modalities). Depending on the task, we may employ a matching loss that forces the embedding vectors between related images and text to be similar, as shown in Figure 30.9. (See Chapter [ch01] for more on embedding vectors.)

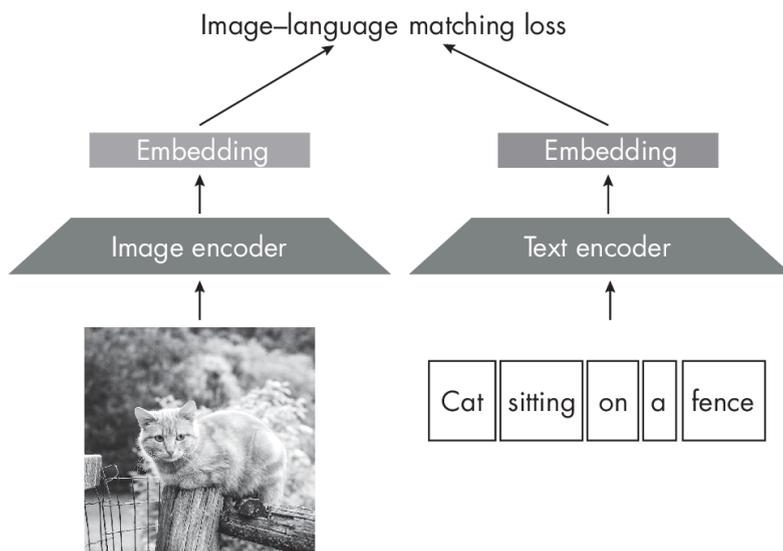


Figure 30.9

Figure 30.9 shows image and text encoders as separate components. The image encoder can be a convolutional backbone or a vision transformer, and the language encoder can be a recurrent neural network or language transformer. However, it's common nowadays to use a single transformer-based module that can simultaneously process image and text data. For example, the VideoBERT model has a joint module that processes both video and text for action classification and video captioning.

Optimizing a matching loss, as shown in Figure 30.9, can be useful for learning embeddings that can be applied to various tasks, such as image classification or summarization. However, it is also possible to directly optimize the target loss, like classification or regression, as Figure 30.10 illustrates.

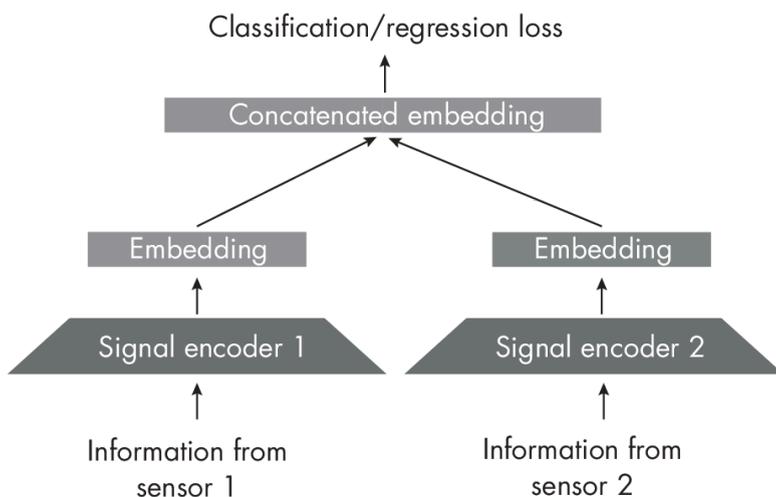


Figure 30.10



Figure [30.10](#) shows data being collected from two different sensors. One could be a thermometer and the other could be a video camera. The signal encoders convert the information into embeddings (sharing the same number of dimensions), which are then concatenated to form the input representation for the model.

Intuitively, models that combine data from different modalities generally perform better than unimodal models because they can leverage more information. Moreover, recent research suggests that the key to the success of multimodal learning is the improved quality of the latent space representation.

Inductive Biases

Choosing models with stronger inductive biases can help lower data requirements by making assumptions about the structure of the data. For example, due to their inductive biases, convolutional networks require less data than vision transformers, as discussed in Chapter [\[ch13\]](#).

Recommendations

Of all these techniques for reducing data requirements, how should we decide which ones to use in a given situation?

Techniques like collecting more data, data augmentation, and feature engineering are compatible with all the methods discussed in this chapter. Multi-task learning and multimodal inputs can also be used with the learning strategies outlined here. If the model suffers from overfitting, we should also include techniques discussed in Chapters [\[ch05\]](#) and [\[ch06\]](#).

But how can we choose between active learning, few-shot learning, transfer learning, self-supervised learning, semi-supervised learning, and weakly supervised learning? Deciding which supervised learning technique(s) to try is highly context dependent. You can use the diagram in Figure [30.11](#) as a guide to choosing the best method for your particular project.

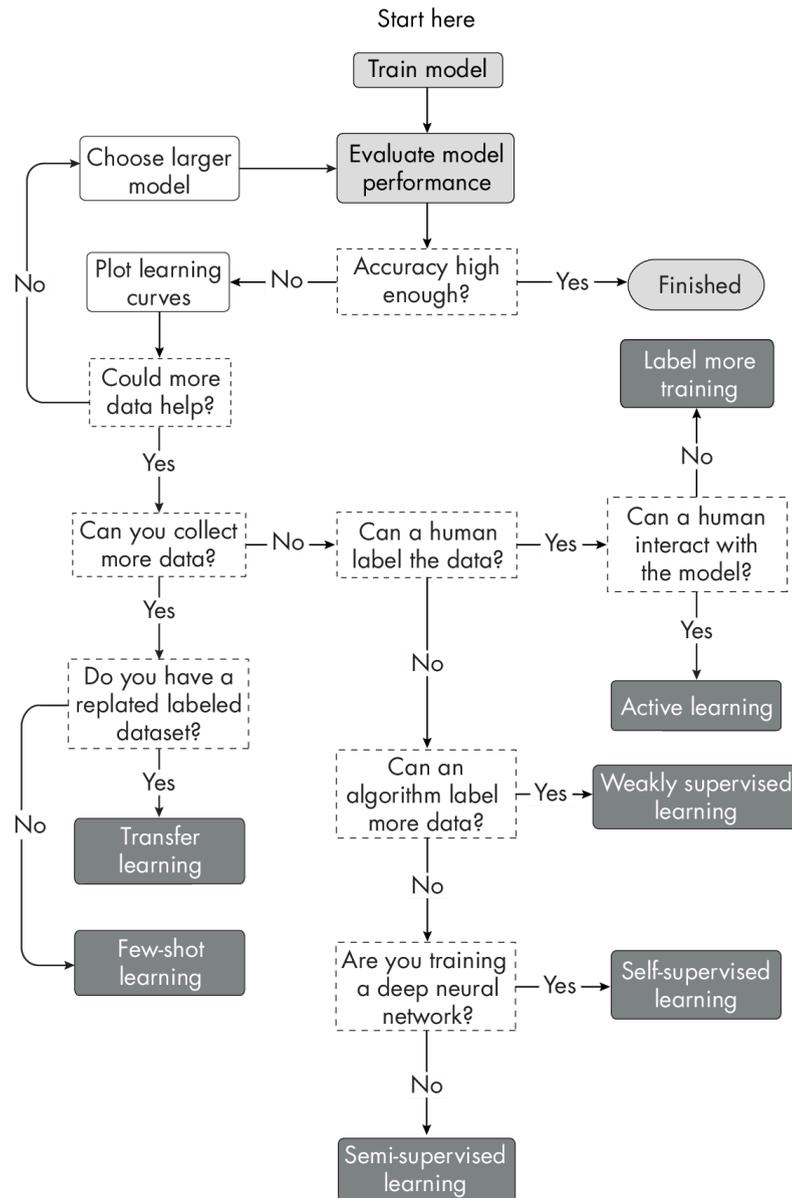


Figure 30.11

Note that the dark boxes in Figure 30.11 are not terminal nodes but arc back to the second box, "Evaluate model performance?"; additional arrows were omitted to avoid visual clutter.

Exercises

30-1. Suppose we are given the task of constructing a machine learning model that utilizes images to detect manufacturing defects on the outer shells of tablet devices similar to iPads. We have access to millions of images of various computing devices, including smartphones, tablets, and computers, which are not labeled; thousands of labeled pictures of smartphones depicting various types of



damage; and hundreds of labeled images specifically related to the target task of detecting manufacturing defects on tablet devices. How could we approach this problem using self-supervised learning or transfer learning?

30-2. In active learning, selecting difficult examples for human inspection and labeling is often based on confidence scores. Neural networks can provide such scores by using the logistic sigmoid or softmax function in the output layer to calculate class-membership probabilities. However, it is widely recognized that deep neural networks exhibit overconfidence on out-of-distribution data, rendering their use in active learning ineffective. What are some other methods to obtain confidence scores using deep neural networks for active learning?

References

- While decision trees for incremental learning are not commonly implemented, algorithms for training decision trees in an iterative fashion do exist: https://en.wikipedia.org/wiki/Incremental_decision_tree.
- Models trained with multi-task learning often outperform models trained on a single task: Rich Caruana, "Multitask Learning"? (1997), <https://doi.org/10.1023%2FA%3A1007379606734>.
- A single transformer-based module that can simultaneously process image and text data: Chen Sun et al., "VideoBERT: A Joint Model for Video and Language Representation Learning"? (2019), <https://arxiv.org/abs/1904.01766>.
- The aforementioned research suggesting the key to the success of multimodal learning is the improved quality of the latent space representation: Yu Huang et al., "What Makes Multi-Modal Learning Better Than Single (Provably)"? (2021), <https://arxiv.org/abs/2106.04538>.
- For more information on active learning: Zhen et al., "A Comparative Survey of Deep Active Learning"? (2022), <https://arxiv.org/abs/2203.13450>.
- For a more detailed discussion on how out-of-distribution data can lead to overconfidence in deep neural networks: Anh Nguyen, Jason Yosinski, and Jeff Clune, "Deep Neural Networks Are Easily Fooled: High Confidence Predictions for Unrecognizable Images"? (2014), <https://arxiv.org/abs/1412.1897>.

