

What does GPT in ChatGPT stand for?



GPT

<u>G</u>enerative

 Can develop coherent and contextually relevant text based on a given prompt

Pre-Trained

 Model has been trained on and learnt from a large amount of data during a pre-training phase

<u>T</u>ransformer

 Based on the transformer architecture

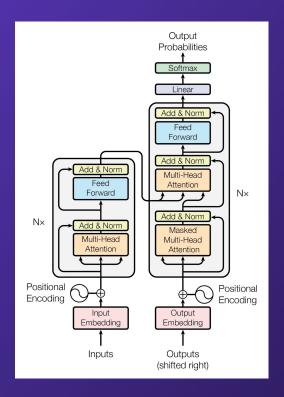


Introduction to Transformers



Introduction to Transformer

- Introduced in the 2017 paper
 "Attention is All You Need"
- Key innovation: the self-attention mechanism
- Self-attention allows the model to weigh the importance of different words in a sentence simultaneously
- No need for sequential processing, enabling massive parallelization





Encoder-Decoder Transformer

- ➤ Add & Norm Softmax Feed Forward Linear Decoder #2 → Add & Norm Multi-Head Add & Norm ◀ Attention Feed Forward - ➤ Add & Norm Add & Norm ≺-Multi-Head Feed Forward Encoder #1 ,--- → Add & Norm Add & Norm ≺-Multi-Head Multi-Head Attention Attention Positional Positional Encodina Encodina Output Input Embedding Embedding Previous Outputs Inputs (shifted right)

Output Probabilities

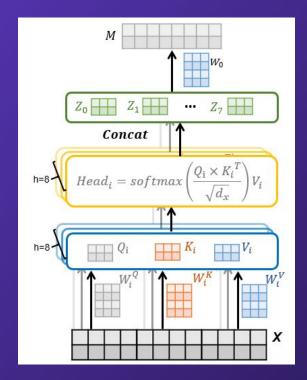
Encoder

Decoder



Multi-Head Self-Attention

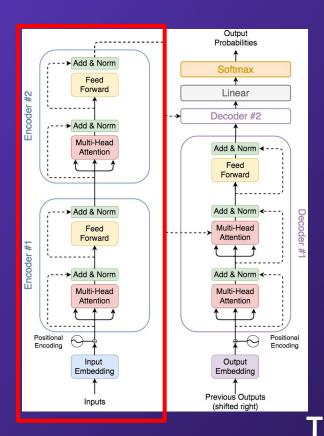
- The heart of the transformer
- Each word in the sentence attends to all other words, determining relationships and importance
- Multi-head means multiple sets of attention calculations run in parallel, each focusing on different aspects of the input





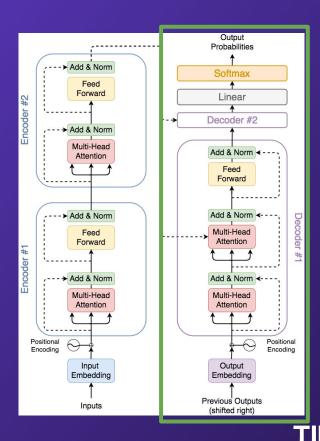
Encoder Blocks

- The encoder consists of:
 - Input Embeddings
 - Positional Encoding
 - Attention blocks that include:
 - Multi-head self-attention layer
 - Feed-forward network for further processing
 - Add & normalize layers for stabilizing training



Decoder Blocks

- The decoder has a similar structure to the encoder but with an extra element
- Masked multi-head attention: Ensures the model only looks at past words when generating text



Large Language Models (LLMs)

- Large Language Models (LLMs) like GPT-3 and T5 form the backbone of zero/few-shot learning
- Pre-trained on massive text corpora, LLMs acquire rich linguistic representations and implicit knowledge





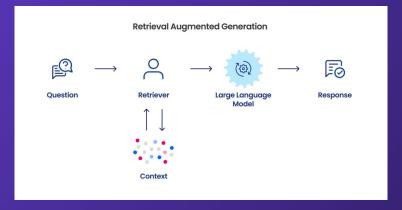
Few-Shot & Zero-Shot

- Few-Shot Learning:
 - Aims to learn from a very small number of labeled examples
 - Leverages prior knowledge from pre-training on broader tasks
 - Better at adapting to new tasks with minimal data
- Zero-Shot Learning:
 - Adapts to a new task without any explicit training examples
 - Relies heavily on pre-trained models' (ex: LLMs) ability to understand language and extract information from prompts and instructions



Zero/Few-Shot QA Techniques

- Prompt Engineering: Carefully designing prompts that best leverage the LLM's capabilities
- Answer Verification: Adding mechanisms to assess the quality and reliability of generated answers
- Retrieval Augmented Generation: Incorporating external knowledge bases or structured data to enhance LLM reasoning





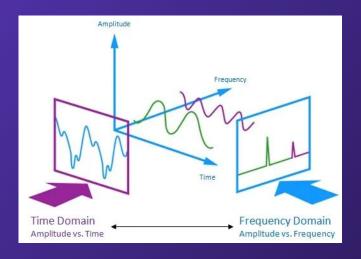
Hands-on activity

Advanced Audio Processing



Importance of Feature Extraction

- Raw audio signals are complex and contain a mix of relevant and irrelevant information
- Feature extraction transforms raw audio into compact, meaningful representations that highlight speech patterns
- Effective features make it easier for ASR models to distinguish between different sounds and words
- The quality of feature extraction directly impacts the overall accuracy of an ASR system





From Time to Frequency Domain

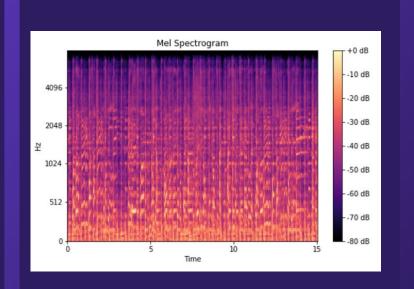
 ASR systems rely heavily on transforming speech signals from time domain to frequency domain to reveal the underlying frequencies that make up the speech

- Time domain: Speech is represented as a waveform where the amplitude (intensity) of the signal is plotted over time
- Discrete Fourier Transform (DFT) is used to decompose a time-domain signal into individual frequencies and amplitudes
- The output of the DFT tells us how much energy is present at each frequency in the original speech signal

Time Domain

Mel-Spectrograms

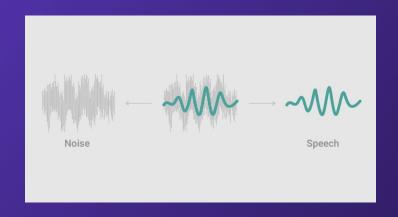
- Type of spectrogram where the frequency scale is converted to the Mel scale
- Mel scale more closely approximates human auditory system's response than the linear frequency scale; making it more effective for audio-related tasks in human speech and music





Noise Reduction

- Noise can come from various sources: background conversations, traffic, machinery, wind, etc
- Noise masks important speech features and disrupts the acoustic signal
- This noise makes it harder for ASR systems to identify words and phrases correctly
- Noise reduction helps minimize background noise without distorting speech





Noise Reduction Techniques

- Spectral Subtraction
 - Estimates the noise spectrum during non-speech segments (silent periods)
 - Subtracts the estimated noise spectrum from the speech spectrum
 - Effective for broadband noise, but it can introduce artifacts if the noise is not stationary or if it overlaps with speech frequencies



Noise Reduction Techniques

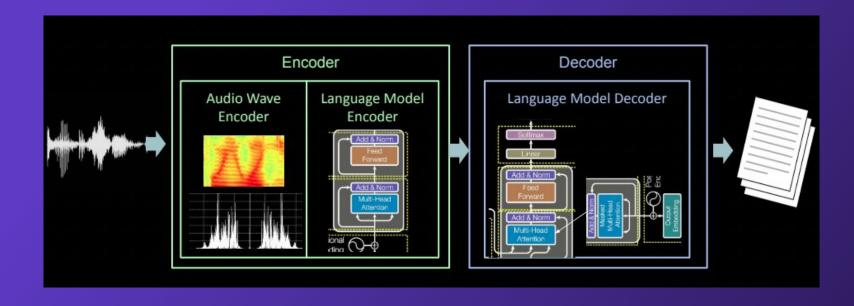
- Deep Neural Networks-Based
 - Neural Networks can be trained to distinguish between speech and noise, enhancing speech recognition accuracy
 - Can learn complex relationships between speech and noise, allowing for highly adaptive noise reduction in various scenarios
 - Example: <u>Deep Denoising Convolutional Neural Network (DnCNN)</u>



Transformers in ASR



How Transformers Process Speech





How Transformers Process Speech

 Speech as input is first converted into a spectrogram or a sequence of feature vectors, which represent the audio signal's power at various frequencies over time

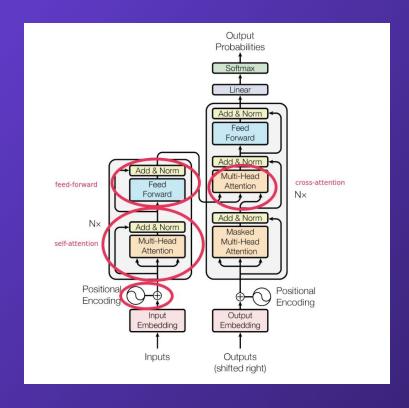
Encoder:

- The encoder processes the speech input using multi-headed self-attention, allowing the model to focus on different parts of the speech input concurrently
- This helps recognize patterns like phonemes, syllables, and words by analyzing features in parallel

Decoder:

 The decoder receives acoustic information from the encoder via cross-attention and combines it with a causal self-attention mechanism to predict the final transcript.

How Transformers Process Speech



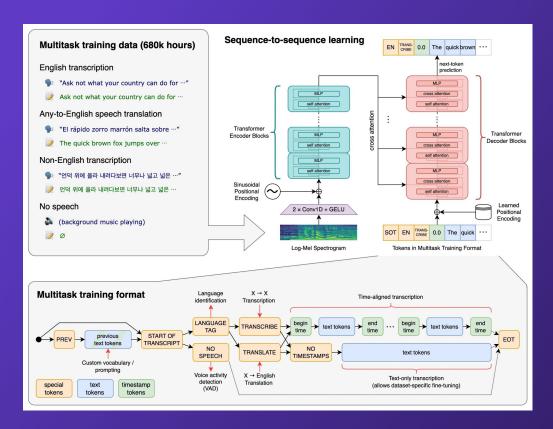


Whisper Architecture

- Core Concept: Encoder-Decoder Transformer
- Key Components:
 - Audio Preprocessing
 - Encoder
 - Decoder
 - Special Tokens and Multitasking



Whisper Architecture





Audio Preprocessing

- Divides the raw input audio into 30-second chunks for efficient processing
- Each chunk is then converted into a log-Mel spectrogram
- Whisper can translate non-English audio directly into English text, having learned bilingual mappings during training on multilingual datasets



Hands-on activity