Audio Fingerprinting

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Tutorial Outline



Introduction



Fingerprinting



Indexing



Literature Review

Introduction



What's this song?



Audio fingerprinting

- Generates content-based summary of an audio signal.
- Allows efficient storage and retrieval of similar audio in large database.

Applications







Music Identification Broadcast monitoring

Second screen applications

Challenges

Robustness

against audio distortions such as noise and reverberation



Search speed

to expedite the retrieval process to enhance user experience



Computational viable

for practical deployment



Modules





Audio Processing

Representation Learning

Indexing

C.

Includes audio segmentation and pre-processing

Generates compact and robust audio representations

Enables fast retrieval

Fingerprinting

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01 Audio Processing

- Waveform: A digitized audio signal.
- **Preprocessing:** Resampling, filtering, audio normalization
- **Time-frequency representation:** 2-D matrix representation of frequency contents of an audio signal over time.
 - Spectrograms and its variants
 - Very few audio fingerprinting approaches directly process waveforms



source: https://jvbale n.github.io/n otes/wavefor m.html



02 Representation Learning

Transform audio segments into low-dimensional vectors.

Rule-based

- Handcrafted features
- Not resilient against high distortion environments.
- Exemplar:
 - o <u>Shazam</u>: peak based
 - <u>Philips</u>: energy-difference-based
 - <u>Waveprint</u>: top-k wavelets

Learning-based

- Real-valued features
- Highly resilient
- Exemplar:
 - o <u>RNN-based</u>
 - o <u>CNN-based</u>
 - o <u>Transformer-based</u>

02 Deep embeddings

- Deep neural networks f(.) as nonlinear embedding function.
- Gradient descent optimization
- Deep embeddings ↔ fingerprints



02 Self-supervised learning





 $oldsymbol{x}_i$ and $oldsymbol{x}_j$ are relevant, not $oldsymbol{x}_k$

02 Self-supervised learning

- Input: log Mel spectrogram (segment): x_i
- Model: CNN
- Training: Contrastive learning with
 - Distorted audio: x_j
 - Any other audio: x_k



Contrastive Loss

02 Contrastive Training

- Distortions are randomly applied:
 - Real-world noises: 0-25 dB SNR
 - \circ Reverberation: RIRs with t₆₀ levels ranging from 0.1s 0.8s
 - **Time offset:** 50 ms

Coding time!





Indexing



03 Retrieval

addillate addillate addillate





MATCHING ALGORITHM



03 Retrieval



- N D-dim database vectors
- Task: For the given query, find the closest vector from the database
- Linear scan: O(ND), slow
- ANN: Sublinear query time
 - don't necessarily have to be exact neighbors
 - Trade off: runtime, accuracy and memory-consumption

 ANN algorithms: LSH (hash-based), PQ (vector quantization-based), HNSW (graph based)

Indexing



03 Data

• Music: Free Music Archive

dataset	clips	genres	length	size	
			[s]	[GiB]	#days
small	8,000	8	30	7.4	2.8
medium	25,000	16	30	23	8.7
large	106,574	161	30	98	37
full	106,574	161	278	917	343

Michaël Defferrard, Kirell Benzi, Pierre Vandergheynst, and Xavier Bresson. Fma: A dataset for music analysis. arXiv preprint arXiv:1612.01840, 2016.

03 Evaluation

- Coarse search: Audio file with maximum matching segments
- Fine-grained search:
 - start end matching times
 - find candidate sequence of segments
 - edit distance

• Metric:

$$\circ \text{ recall}@top-k = \frac{n \text{ hits } @ \text{ top-k}}{n \text{ hits } @ \text{ top-k} + m \text{ miss } @ \text{ top-k}}$$

$$\circ$$
 MRR = $\frac{1}{|Q|} \sum_{i=1}^{|Q|} \frac{1}{rank(q_i)}$



Coding time!



Literature Review



04 Shazam

- Identifies frequencies of peak intensity
- Generates pairs of anchor and target peaks and their corresponding time difference.
- Hashes each pair into a hash table.
- Hash table lookup for query matching.



04 ML approaches

 Enhances interesting spatial patches by assigning more weight to time indices and frequency bands containing salient patches

$$egin{aligned} a^{temp} &= ext{softmax}(X^TW_{temp})\ a^{spect} &= ext{softmax}(X^TW_{spect})\ A &= a^{temp} \otimes a^{spect} imes s\ X' &= A \odot X \end{aligned}$$

Singh, Anup, Kris Demuynck, and Vipul Arora. "Attention-based audio embeddings for query-by-example." *arXiv preprint arXiv:2210.08624* (2022).



Layer	Input size	Output size	
Encoder:			
CNN layer	$1 \times 64 \times 96$	32×64×96	
ResBlock1	32×64×96	32×64×96	
ResBlock2	32×64×96	64×32×48	
ResBlock6	512×4×6	$1024 \times 2 \times 3$	
Flatten		6144	
Projection Head:	d * i	d * o	
Conv1D + ELU	128×48	128×32	
Conv1D	128×32	128×1	

04 ML approaches

- CNN architecture with spatially separable convolutions layers.
- Divide-and-encode: splits embedding into chunks
- Triplet loss



04 LSH

- Map similar samples to same hash code
- K random hyperplanes: h_1 , h_2 , ... $h_K \Rightarrow 2^K$ disjoint partitions of space.
- K-bit hash code for sample **a**: Step $[a^{T}h_{1}a^{T}h_{2}]_{a}a^{T}h_{K}]$
- Exact comparison of a with points mapped to same hash bucket - May miss near neighbors
- Repeat L times w.r.t different set of K hyperplanes.
- Distance computation with candidates mapped to same hash code as a across L tables.



Lv, Qin, et al. "Multi-probe LSH: efficient indexing for high-dimensional similarity search." Proceedings of the 33rd international conference on Very large data bases. 2007.

04 LSH

- Computational Cost:
 - N points, D dimensional, K hyperplanes
 - DK: generate K-bit hash code ⇒ Mapping sample to a hash bucket. Cost of dot product of a sample with K hyperplanes.
 - Assume on average, each bucket contains: N/2^K
 - Exact comparison cost: DN/2^K
 - Repeat everything L times (no. of hash tables)
 - Cost: $LDK + LDN/2^{K} \rightarrow O(\log N)$, if K = log N



04 Recent Works

ATTENTION-BASED AUDIO EMBEDDINGS FOR QUERY-BY-EXAMPLE

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ABSTRACT

arXiv:2210.08624v1 [eess.AS] 16 Oct 2022

An ideal audio retrieval system efficiently and robustly recognizes a short query snippet from an extensive database. However, the performance of well-known audio fingerprinting systems falls short at high signal distortion levels. This paper presents an audio retrieval system that generates noise and reverberation robust audio fingerprints using the contrastive learning framework. Using these fingerprints, the method performs a comprehensive search to identify the query audio and precisely estimate its timestamp in the reference audio. Our framework involves training a CNN to maximize the similarity between pairs of embeddings extracted from clean audio and its corresponding distorted and time-shifted version. We employ a channelwise spectral-temporal attention mechanism to better discriminate the audio by giving more weight to the salient spectral-temporal patches in the signal. Experimental results indicate that our system is efficient in computation and memory usage while being more accurate, particularly at higher distortion levels, than competing state-of-the-art systems and scalable to a larger database.

1. INTRODUCTION

Audio fingerprinting is the principal component of an audio identification task. Finding perceptually similar audio in a massive audio corpus is computationally and memory expensive. Audio fingerprinting is a technique that derives a content-based audio summary and links it with similar audio fragments in the database. It allows for an efficient and quick search against other audio fragments. There are several possibilities for fingerprinting applications on digital devices, such as smartphones and TVs, that are becoming ubiquitous. Music identification on moled devices, used on query-by-example, is a common use case in which a user hears a song in a public area and wants additional information about it [11.2]. The second-

NEURAL AUDIO FINGERPRINT FOR HIGH-SPECIFIC AUDIO RETRIEVAL BASED ON CONTRASTIVE LEARNING

work, e

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ABSTRACT

Most of existing audio fingerprinting systems have limitations to be used for high-specific audio retrieval at scale. In this work, we generate a low-dimensional representation from a short unit seement of segmen audio, and couple this fingerprint with a fast maximum inner-product att search. To this end, we present a contrastive learning framework that derives from the segment-level search objective. Each update in training uses a batch consisting of a set of pseudo labels, randomly selected original samples, and their augmented replicas. These replicas can simulate the degrading effects on original audio signals by fingerapplying small time offsets and various types of distortions, such printer as background noise and room/microphone impulse responses. In the segment-level search task, where the conventional audio fingerprinting systems used to fail, our system using 10x smaller storage has shown promising results. Our code and dataset are available at https://mimbres.github.ic/neural-audio-fp/. Fig. 1.

Index Terms— acoustic fingerprint, self-supervised learning, data augmentation, music information retrieval segmen of audi

1. INTRODUCTION

Audio fingerprinting is a content summarization technique that links short singles of unlabeled audio contents to the same contents in the database [1]. The most well-known application is the music fingerfrom the microphone or streaming audio input. Other applications from the microphone or streaming audio input. Other applications from the microphone or streaming audio input. Other applications from the microphone or streaming audio input. Other applications from the microphone or streaming audio input. Other applications from the microphone or streaming audio input. Other applications from the microphone or streaming audio input from the streaming General requirements for audio fingerprinting system are dis-

criminability over a huge number of other fingerprints, robustness • We e

SIMULTANEOUSLY LEARNING ROBUST AUDIO EMBEDDINGS AND BALANCED HASH CODES FOR QUERY-BY-EXAMPLE

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ABSTRACT

Audio fingerprinting systems must efficiently and robustly identify query snippets in an extensive database. To this end, state-of-the-art systems use deep learning to generate compact audio fingerprints. These systems deploy indexing methods, which quantize fingerprints to hash codes in an unsupervised manner to expedite the search. However, these methods generate imbalanced hash codes, leading to their suboptimal performance. Therefore, we propose a self-supervised learning framework to compute fingerprints and balanced hash codes in an end-to-end manner to achieve both fast and accurate retrieval performance. We model hash codes as a balanced clustering process, which we regard as an instance of the optimal transport problem. Experimental results indicate that the proposed approach improves retrieval efficiency while preserving high accuracy, particularly at high distortion levels, compared to the competing methods. Moreover, our system is efficient and scalable in computational load and memory storage.

Index Terms— Audio fingerprinting, optimal transport, hashing efficient retrieval self-supervised learning mapping is prone to being non-discriminative and/or noise-sensitive, thus requiring multiple costly probes of hash buckets to achieve satisfactory performance.

Therefore, simultaneously learning audio embeddings and binary encodings is a promising solution for improving retrieval effectiveness. However, learning binary encoding is a discrete learning process, making end-to-end training challenging. In addition, a uniform distribution (code balance) of binary encodings is required to facilitate ar efficient search in the Hamming space. Existing works, on learned hash functions [11–3] tend to suffer from the code imbance problem, realling in sub-primal (disory retrieval perforbance).

This paper presents an audio fingerprinting system that is robust against high noise and reverberation levels and performs a comprehensive audio search. Overall, the main contributions of our work are as follows:

 An end-to-end self-supervised learning framework for jointly learning continuous and discrete audio embeddings (binary encodings). To our knowledge, no prior work exists for the audio

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Faiss

Faiss is a library for efficient similarity search and clustering of dense vectors. It contains algorithms that search in sets of vectors of any size, up to ones that possibly do not fit in RAM. It also contains supporting code for evaluation and parameter tuning.

Faiss is written in C++ with complete wrappers for Python. Some of the most useful algorithms are implemented on the GPU. It is developed primarily at FAIR, the fundamental AI research team of Meta.

What is similarity search?

Given a set of vectors x_i in dimension d, Faiss builds a data structure in RAM from it. After the structure is constructed, when given a new vector x in dimension d it performs efficiently the operation:

 $j = argmin_i \|x - x_i\|$

For more information ...

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Thank you!