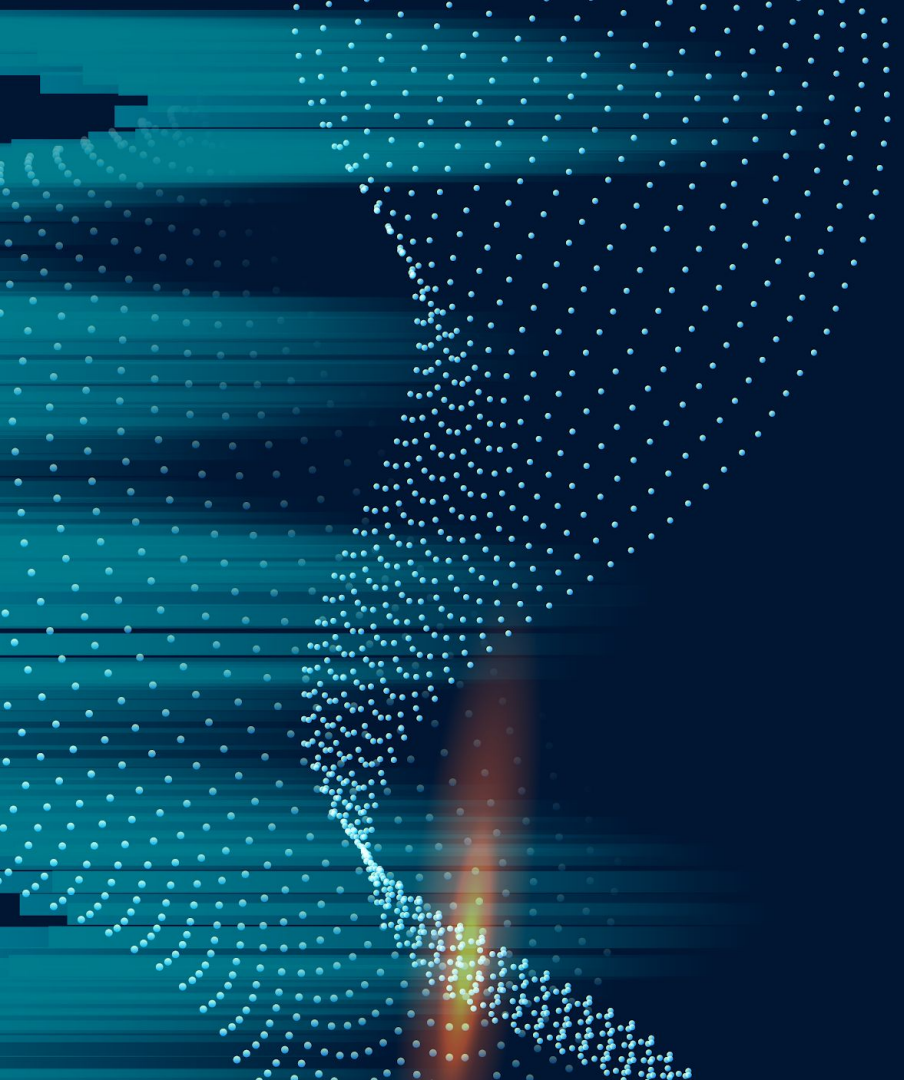
Kaldi ASR Modeling:

Robust Automatic Speech Recognition using Kaldi for Radiotelephonic Applications



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Product Owners

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Coding, Research, Documentation

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Coding, Research, Documentation

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Research, Documentation

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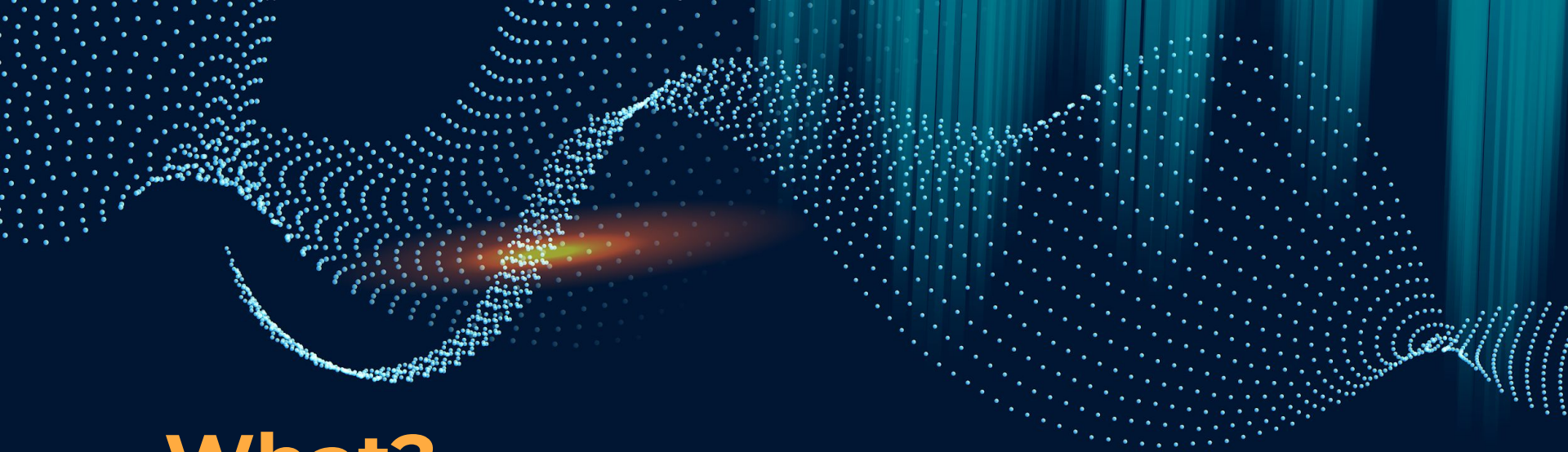
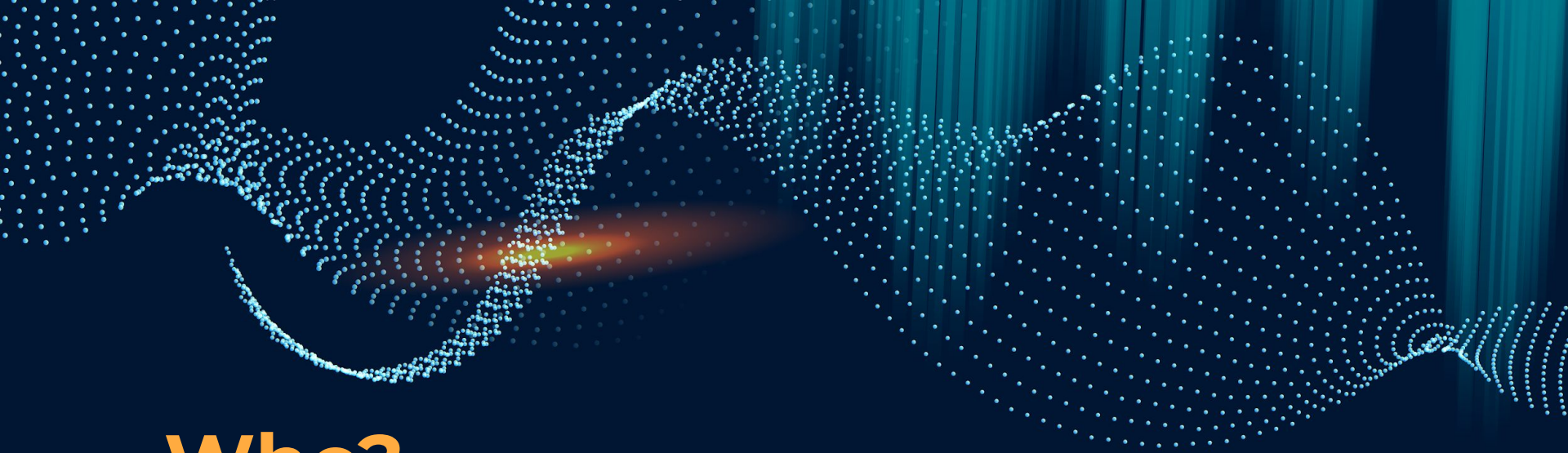


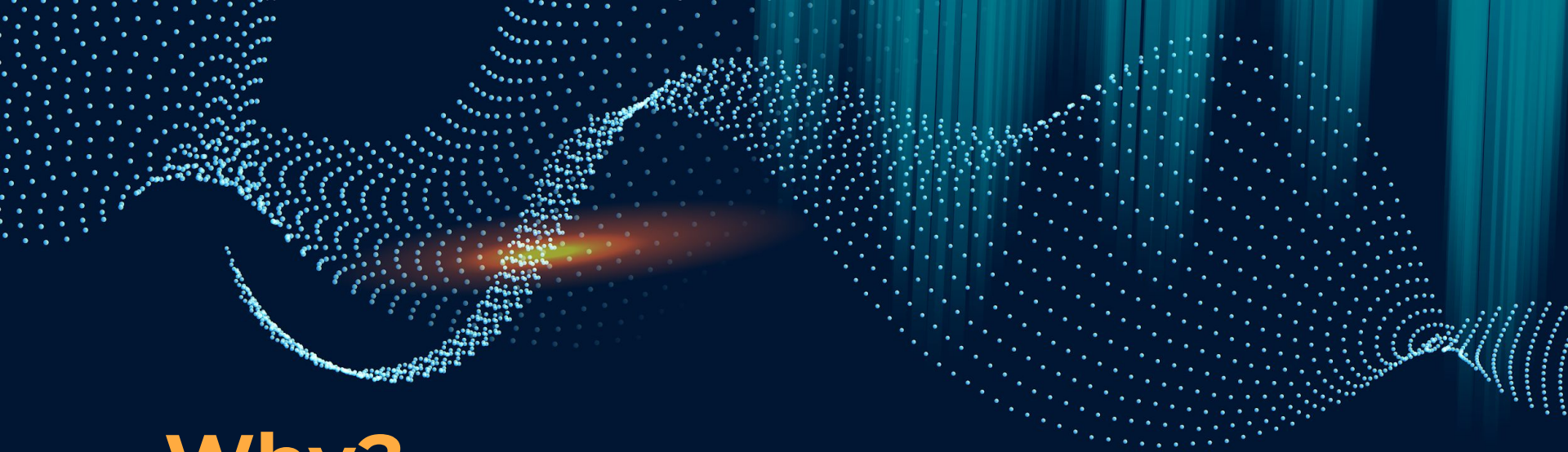
What?

Develop acoustic ASR models capable of transcribing live ATC transmissions in real-time using the Kaldi ASR Toolkit and the ATC02 corpus.



Who?

- Ab initio pilots
- Flight Instructors
- Air Traffic Control Operators
- Hobbyists



Why?

- Miscommunications between ATC and pilots due to a lack of formal training for pilots
- Ability to review communications on the ground after flights for training purposes

End to End vs Acoustic Models

End-to-End (E2E) Models

- Relationship between audio signals and the words
- Large dataset is required
- Larger and more complex models

Acoustic Models

- Relationships between audio signals and the phonetics
- Better results with small dataset
- Smaller models

Design Constraints (Training)

- Minimum storage size of 12.5 GB
- Minimum video memory (VRAM) of 12 GB for model training
  - NVidia GeForce RTX 4080
- AMD processor (recommended)

Design Constraints (Transcribing)

- All inputs as WAV files
  - Convert non-WAV files using FFMPEG utility
- General American English
  - E.g., “color” versus “colour”
- No punctuation or grammatical marks
  - Commas, colons, hyphens, etc.

Assumptions and Dependencies

- Clear and direct communication
  - Low interference and background noise
- General American English
- 20- to 100-hour dataset
- Sufficient storage
- Sufficient video memory
- Operating system
  - Linux (e.g., Ubuntu, Debian, etc.)
  - Windows Subsystem for Linux (WSL)
  - Linux virtual machine for MacOS

System Architecture (Preprocessing)

ATC0 Corpus

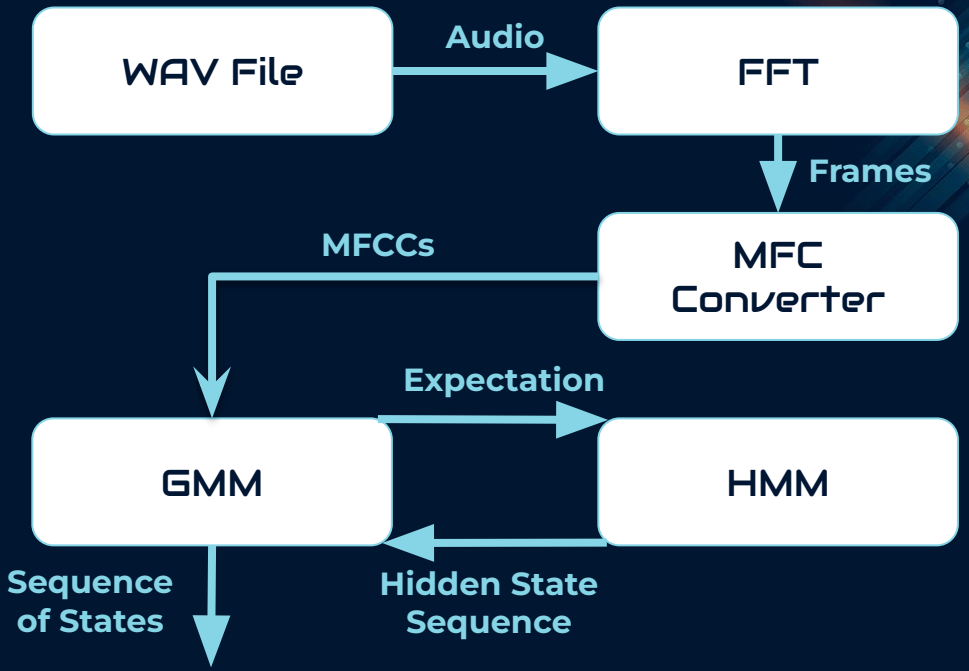
- Linguistic Data Consortium (LDC)
- 30-hour ATC dataset
  - Audio files
  - Text transcriptions

ATCO2 Corpus

- Repository for ASR and NLP research
- Provides preprocessing script written in Bash designed for Kaldi ASR Toolkit

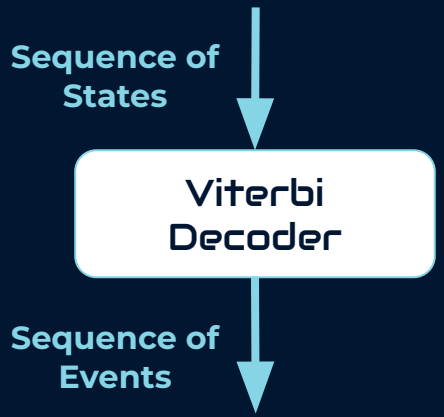
System Architecture (Preparation)

- Setting up audio for decoding
- Fast Fourier Transform (FFT)
  - Time-domain → Frequency domain
- Mel-Frequency Cepstrum (MFC)
  - Mel-Frequency Cepstral Coefficients (MFCCs)
- Gaussian Mixture Model (GMM)
- Hidden Markov Model (HMM)



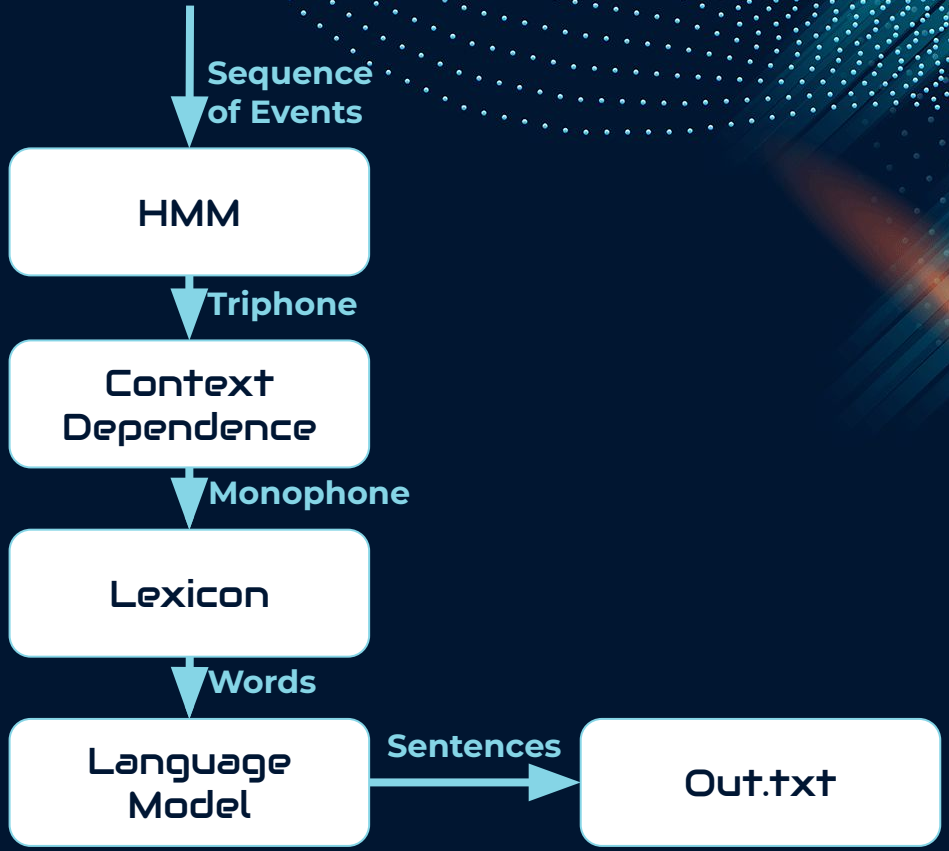
System Architecture (Preparation)

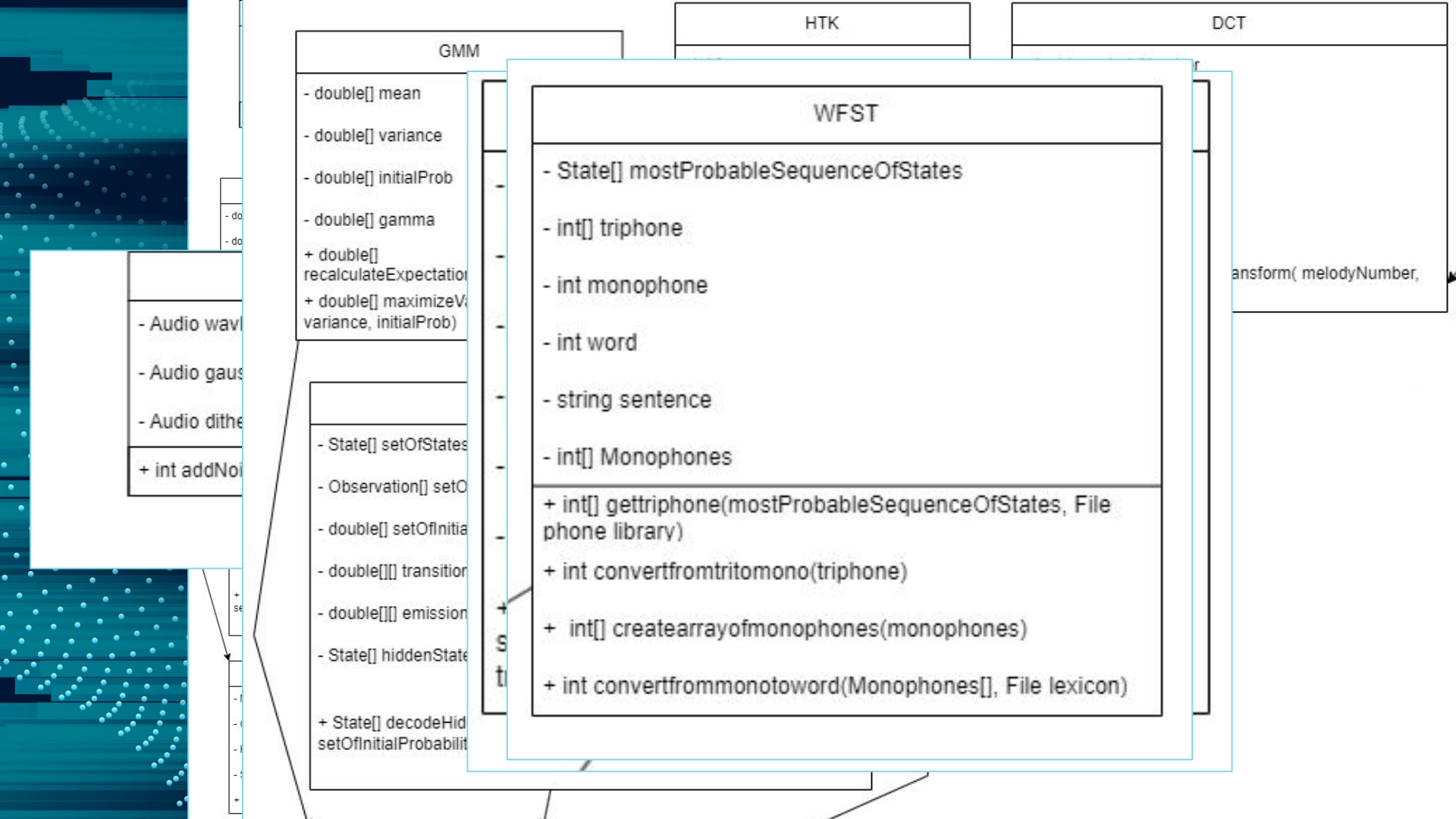
- Using hidden state sequence to find the most probable sequence of phonemes
- Viterbi decoder



System Architecture (Output)

- Converting to sentences
- Weighted Finite State Transducers (WFSTs)
- HMM
- Context Dependence
- Lexicon
- Language Model




HTK
DCT
GMM
- double[] mean
- double[] variance
- double[] initialProb
- double[] gamma
+ double[] recalculateExpectatio
+ double[] maximizeV
variance, initialProb)
ansform(melodyNumber,
- Audio wav
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- Audio dithe
+ int addNoi
- State[] setOfStates
- Observation[] setO
- double[] setOfInitia
- double[][] transitio
- double[][] emission
- State[] hiddenState
+ State[] decodeHid
setOfInitialProbabilit
WFST
- State[] mostProbableSequenceOfStates
- int[] triphone
- int monophone
- int word
- string sentence
- int[] Monophones
+ int[] gettriphone(mostProbableSequenceOfStates, File phone library)
+ int convertfromtritomono(triphone)
+ int[] createarrayofmonophones(monophones)
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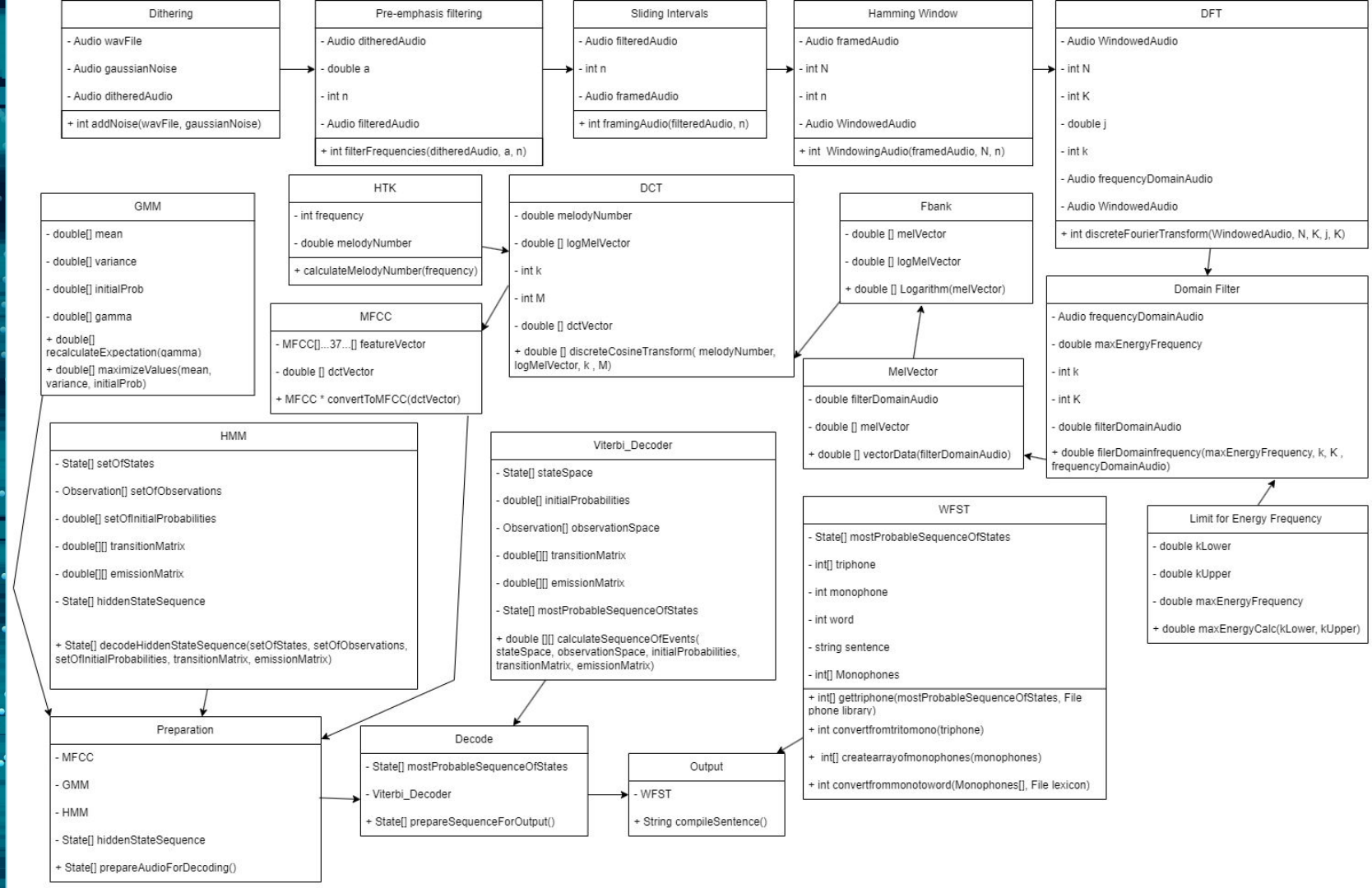
Dithering
- Audio wavFile
- Audio gaussianNoise
- Audio ditheredAudio
+ int addNoise(wavFile, gaussianNoise)
Pre-emphasis filtering
- Audio ditheredAudio
- double a
- int n
- Audio filteredAudio
+ int filterFrequencies(ditheredAudio, a, n)
Sliding Intervals
- Audio filteredAudio
- int n
- Audio framedAudio
+ int framingAudio(filteredAudio, n)
Hamming Window
- Audio framedAudio
- int N
- int n
- Audio WindowedAudio
+ int WindowingAudio(framedAudio, N, n)
DFT
- Audio WindowedAudio
- int N
- int K
- double j
- int k
- Audio frequencyDomainAudio
- Audio WindowedAudio
+ int discreteFourierTransform(WindowedAudio, N, K, j, K)
GMM
- double[] mean
- double[] variance
- double[] initialProb
- double[] gamma
+ double[] recalculateExpectation(gamma)
+ double[] maximizeValues(mean, variance, initialProb)
HTK
- int frequency
- double melodyNumber
+ calculateMelodyNumber(frequency)
DCT
- double melodyNumber
- double [] logMelVector
- int k
- int M
- double [] dctVector
+ double [] discreteCosineTransform(melodyNumber, logMelVector, k , M)
Fbank
- double [] melVector
- double [] logMelVector
+ double [] Logarithm(melVector)
Domain Filter
- Audio frequencyDomainAudio
- double maxEnergyFrequency
- int k
- int K
- double filterDomainAudio
+ double filerDomainfrequency(maxEnergyFrequency, k, K , frequencyDomainAudio)
MFCC
- MFCC[]...37...[] featureVector
- double [] dctVector
+ MFCC * convertToMFCC(dctVector)
MelVector
- double filterDomainAudio
- double [] melVector
+ double [] vectorData(filterDomainAudio)
HMM
- State[] setOfStates
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- double[][] transitionMatrix
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Viterbi_Decoder
- State[] stateSpace
- double[] initialProbabilities
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+ double [][] calculateSequenceOfEvents(stateSpace, observationSpace, initialProbabilities, transitionMatrix, emissionMatrix)
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Limit for Energy Frequency
- double kLower
- double kUpper
- double maxEnergyFrequency
+ double maxEnergyCalc(kLower, kUpper)
Preparation
- MFCC
- GMM
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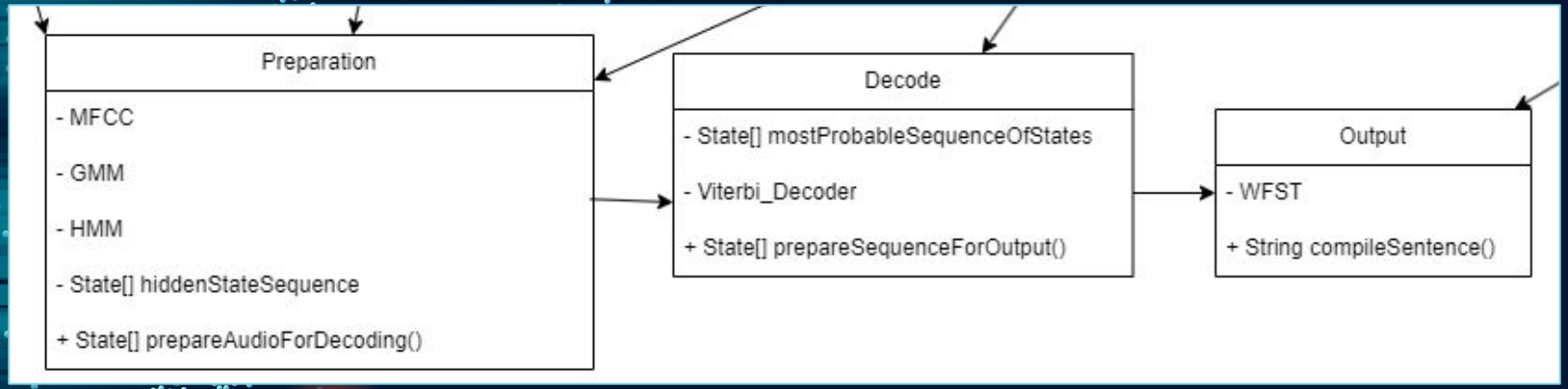
- State[] hiddenStateSequence
+ State[] prepareAudioForDecoding()
Decode
- State[] mostProbableSequenceOfStates
- Viterbi_Decoder
+ State[] prepareSequenceForOutput()
Output
- WFST
+ String compileSentence()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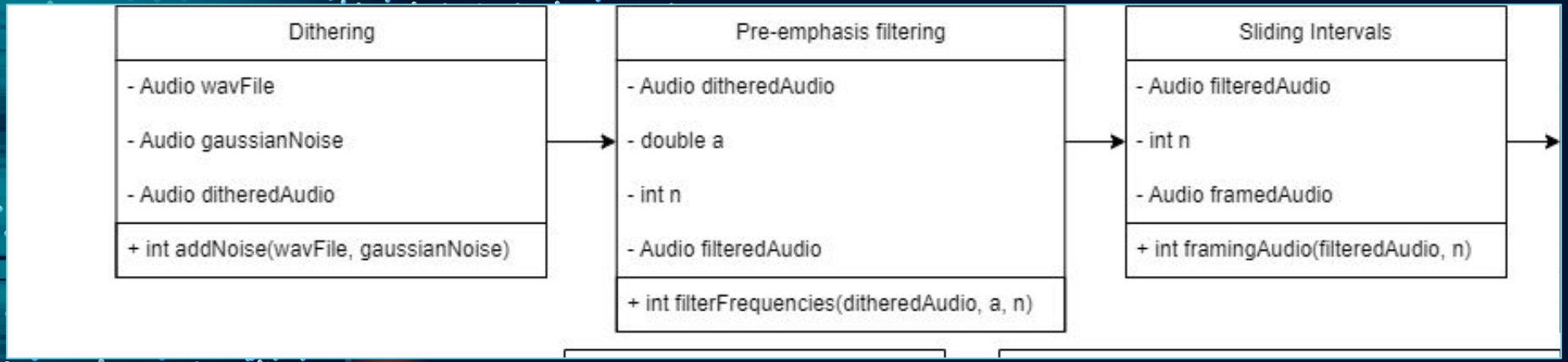
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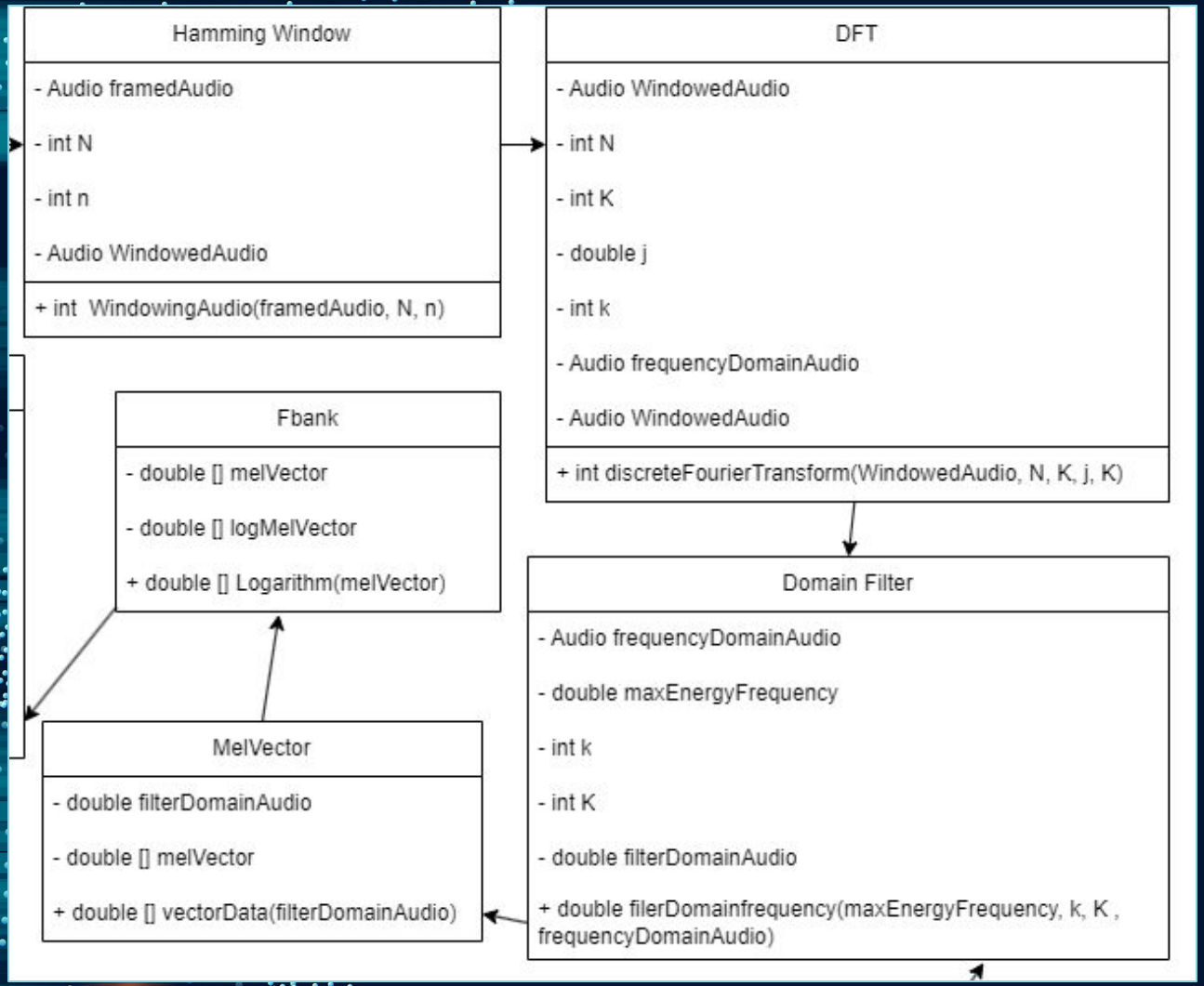
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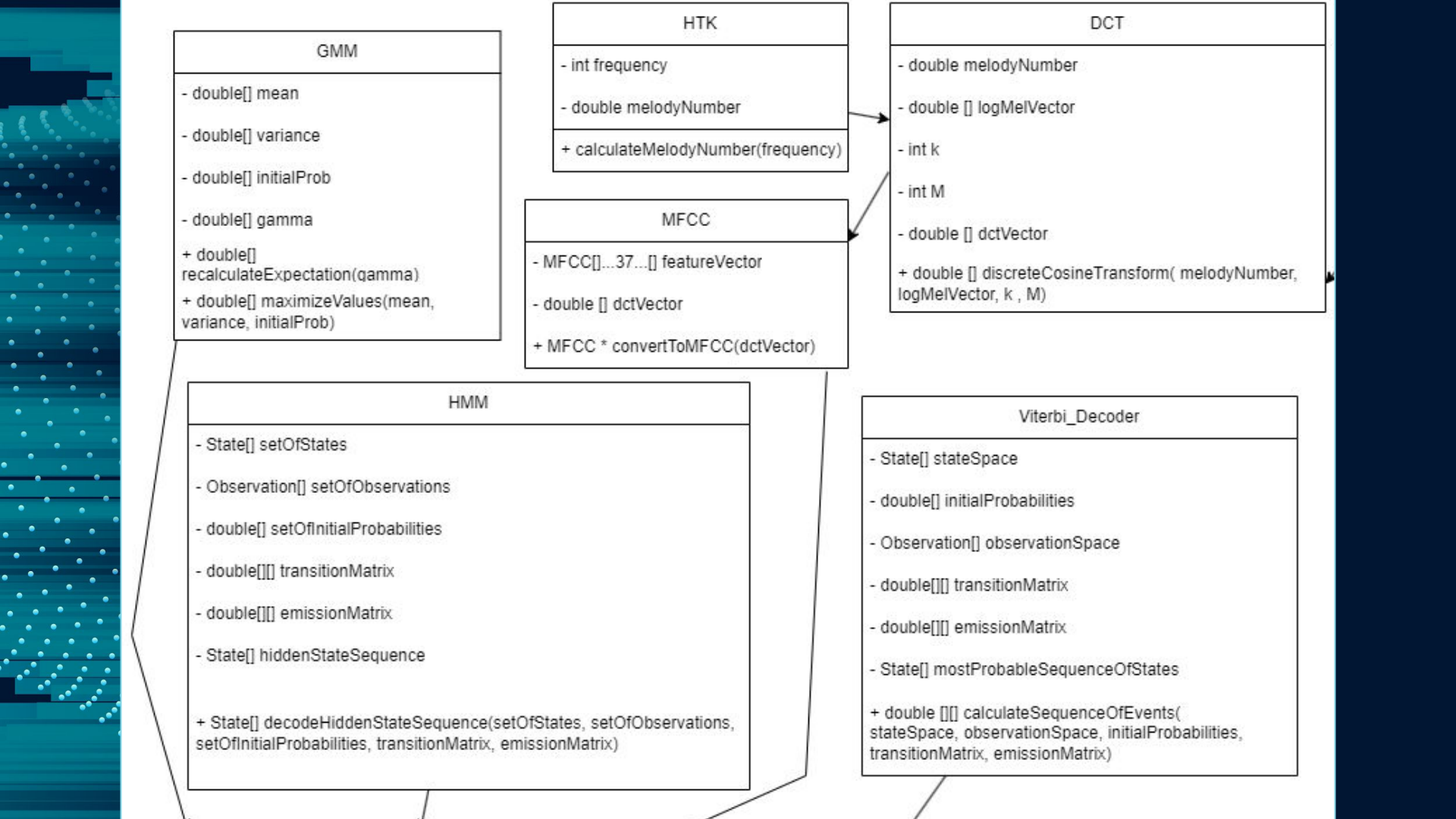
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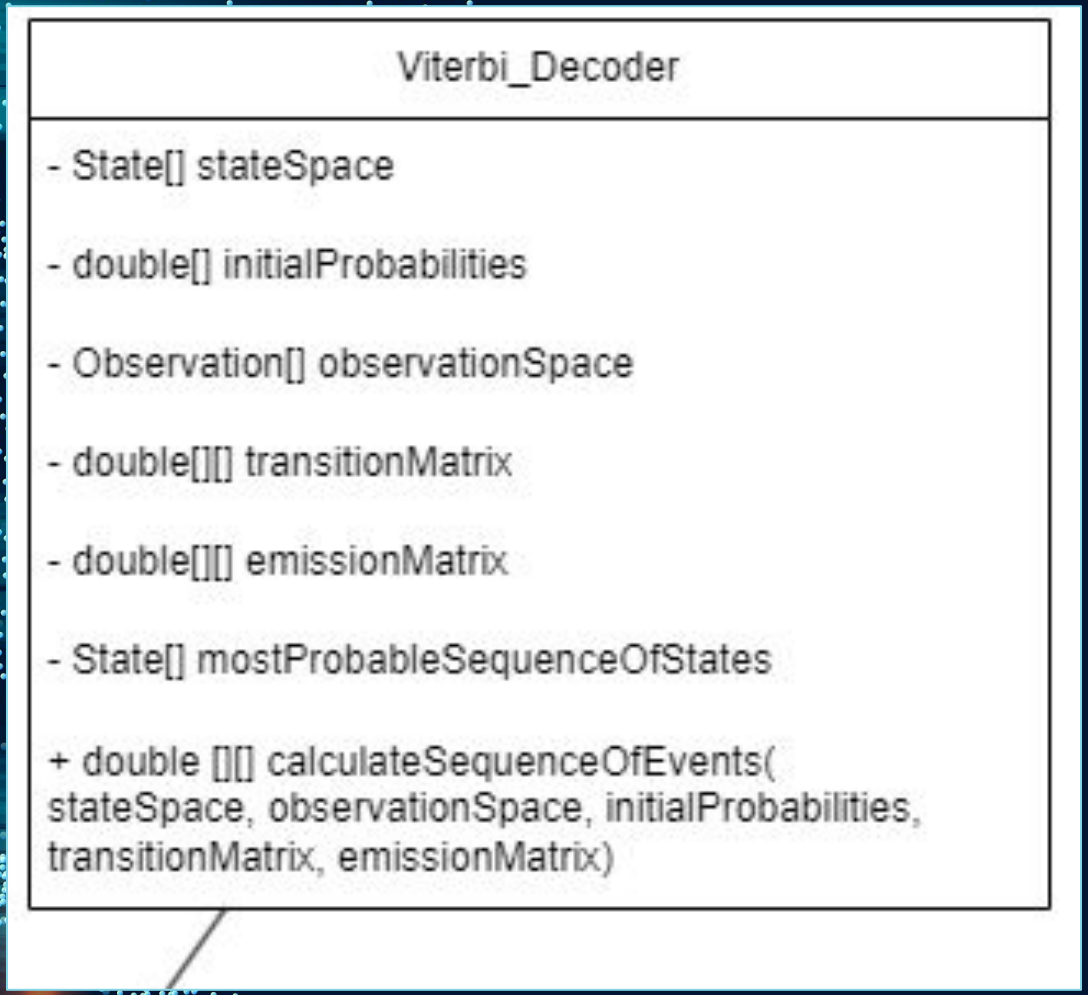
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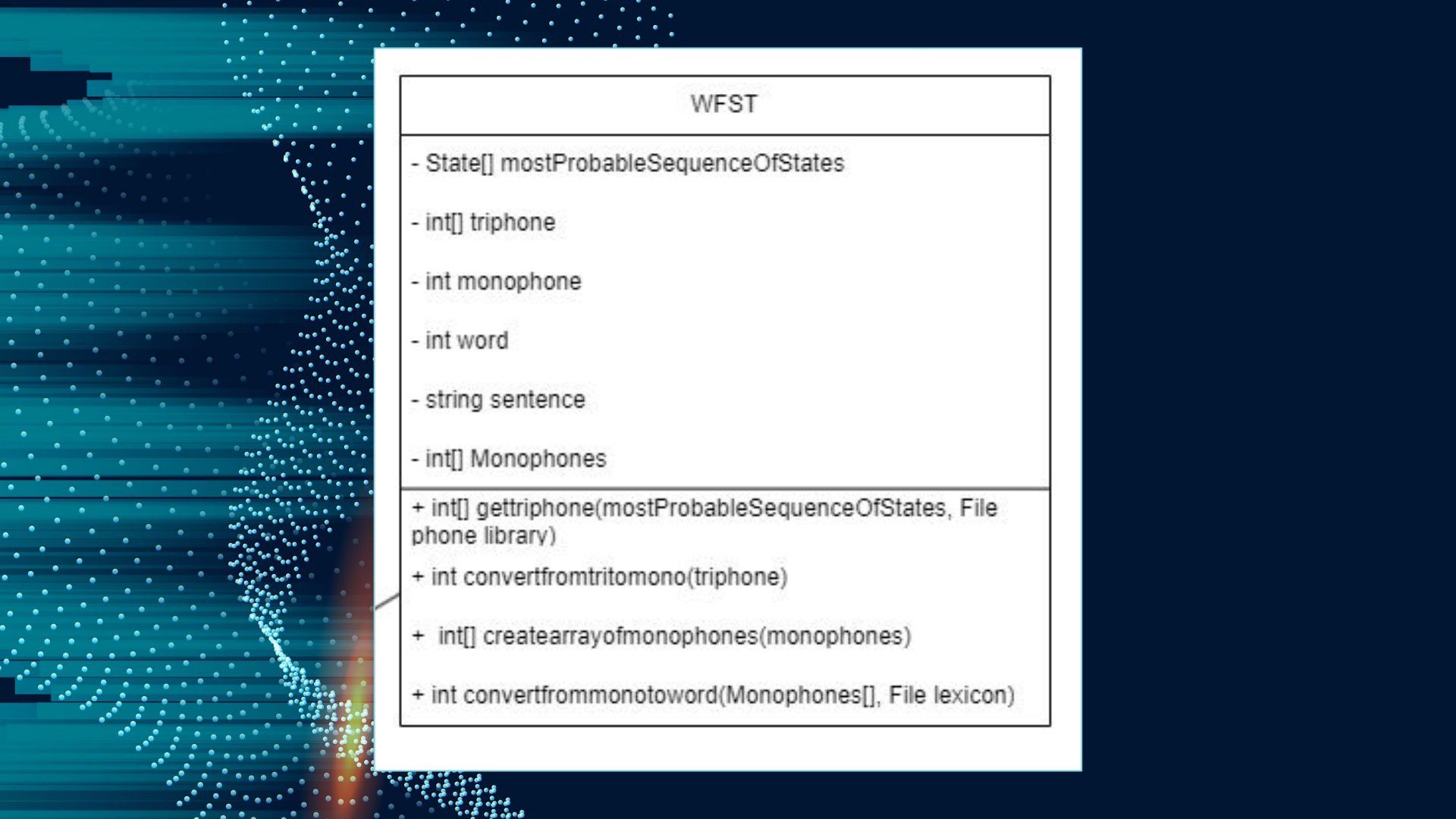
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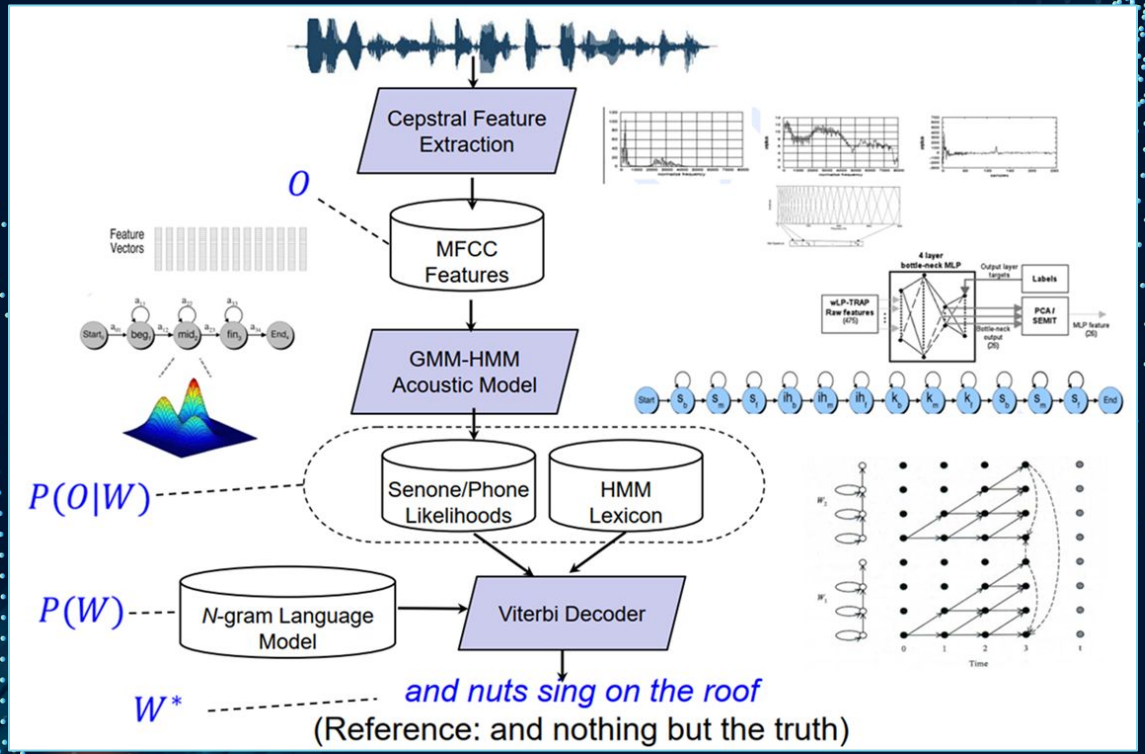
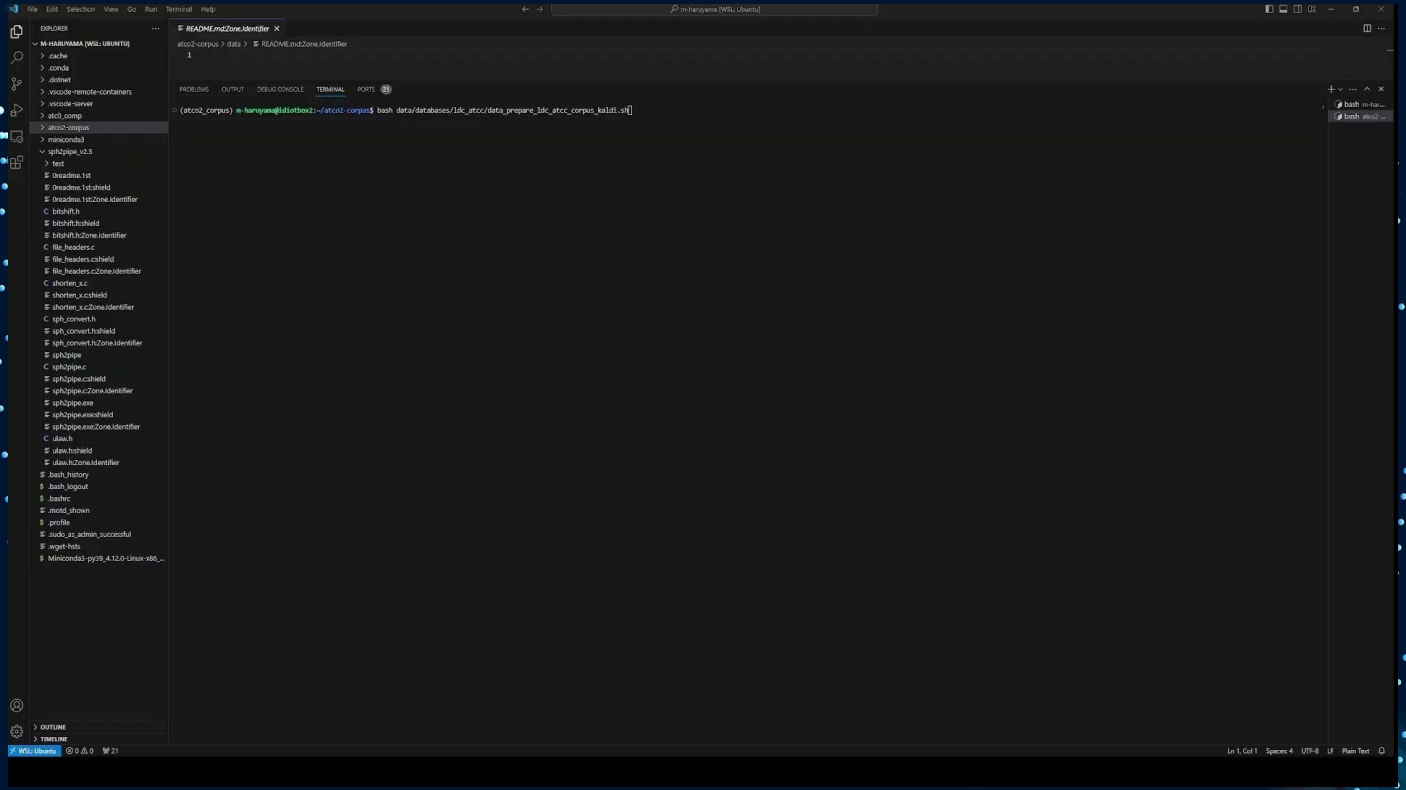
Sub-System Design



Iteration 7.0.0	Iteration 13.1.0	Iteration 17.0.5	Iteration 17.1.0
WER = 18.06%	WER = 13.50%	WER = 13.21%	WER = 13.23%
SER = 78.88%	SER = 72.36%	SER = 71.26%	SER = 72.08%

Demo


File Edit Selection View Go Run Terminal Help
m-haruyama [WSL: Ubuntu]
EXPLORER
README.md:Zone.Identifier
atco2-corpus > data > README.md:Zone.Identifier
1
PROBLEMS OUTPUT DEBUG CONSOLE TERMINAL PORTS 21
(atco2_corpus) m-haruyama@idiotbox2:~/atco2-corpus$ bash data/databases/ldc_atcc/data_prepare_ldc_atcc_corpus_kaldi.sh
bash m-har...
bash atco2-...
M-HARUYAMA [WSL: UBUNTU]
.cache
.conda
.dotnet
.vscode-remote-containers
.vscode-server
atc0_comp
atco2-corpus
miniconda3
sph2pipe_v2.5
test
0readme.1st
0readme.1st:shield
0readme.1st:Zone.Identifier
bitshift.h
bitshift.h:shield
bitshift.h:Zone.Identifier
file_headers.c
file_headers.c:shield
file_headers.c:Zone.Identifier
shorten_x.c
shorten_x.c:shield
shorten_x.c:Zone.Identifier
sph_convert.h
sph_convert.h:shield
sph_convert.h:Zone.Identifier
sph2pipe
sph2pipe.c
sph2pipe.c:shield
sph2pipe.c:Zone.Identifier
sph2pipe.exe
sph2pipe.exe:shield
sph2pipe.exe:Zone.Identifier
ulaw.h
ulaw.h:shield
ulaw.h:Zone.Identifier
.bash_history
.bash_logout
.bashrc
.motd_shown
.profile
.sudo_as_admin_successful
.wget-hsts
Miniconda3-py39_4.12.0-Linux-x86_...
OUTLINE
TIMELINE
WSL: Ubuntu
Ln 1, Col 1 Spaces: 4 UTF-8 LF Plain Text

Q&A