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SPEAKING TO AZURE

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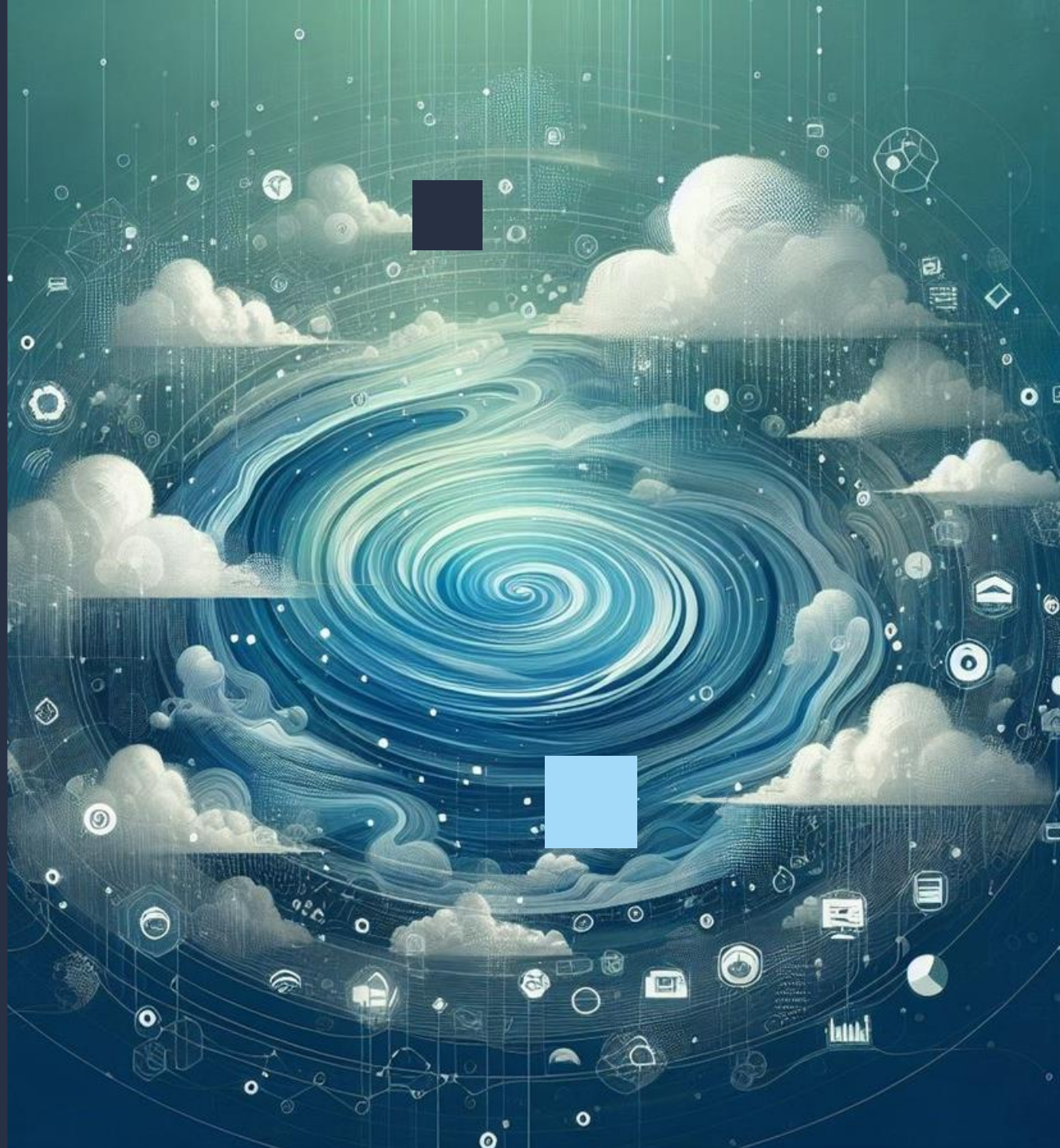

Agenda

- History of speech in Computing
- Speech on Azure
- Video and audio extraction demos

Speech synthesis over the year (pre-Millenium)

1779: **Wolfgang von Kempelen** creates the first known speech synthesis machine, a mechanical device that could simulate simple speech sounds.

1939: **Homer Dudley** of Bell Labs introduces the **Voder**, the first electronic speech synthesizer, at the New York World's Fair.

1950s: **Dudley's Vocoder** (Voice Encoder), another Bell Labs invention, is developed and used for speech compression and transformation.

1961: **John Larry Kelly, Jr.** at Bell Labs uses an IBM 704 to create one of the first computer-generated voices to sing the song "Daisy Bell" ("Bicycle Built for Two").

1980s: The development of **DECtalk**, a speech synthesis system by **Digital Equipment Corporation**, becomes notable for its use in assistive technologies (e.g., used by Stephen Hawking).

1987: **AT&T Bell Labs** releases **Lucent Text-to-Speech (TTS)** system, which is one of the early commercially available text-to-speech systems.

1997: **Microsoft SAPI (Speech Application Programming Interface)** 4.0 is released, allowing developers to create speech-enabled applications for Windows.

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Famous voices:

Sam – Microsoft SAPI 5

Stephen Hawking (Stephen Hawking's Voice Emulator Project | Pawel Wozniak (pawozniak.com)

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Speech synthesis over the year (> 2000)

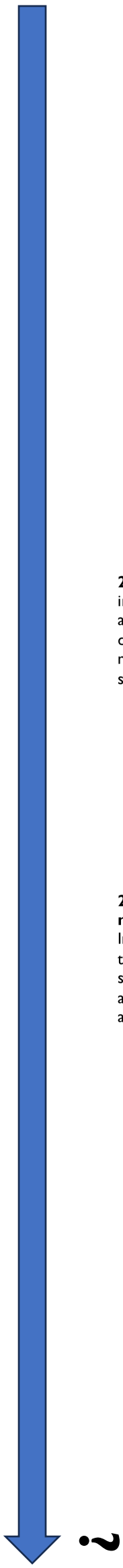
2001: The release of **Festival Speech Synthesis System** by **Centre for Speech Technology Research** (CSTR) at the University of Edinburgh, an open-source framework that becomes popular in academic research.

2010: The development of **neural network-based synthesis** begins to gain traction, with methods like **WaveNet** by **DeepMind** (Google), showing improvements in the naturalness of speech output.

2016: **DeepMind's WaveNet** introduces a groundbreaking approach to speech synthesis using deep learning, producing much more natural and human-like speech.

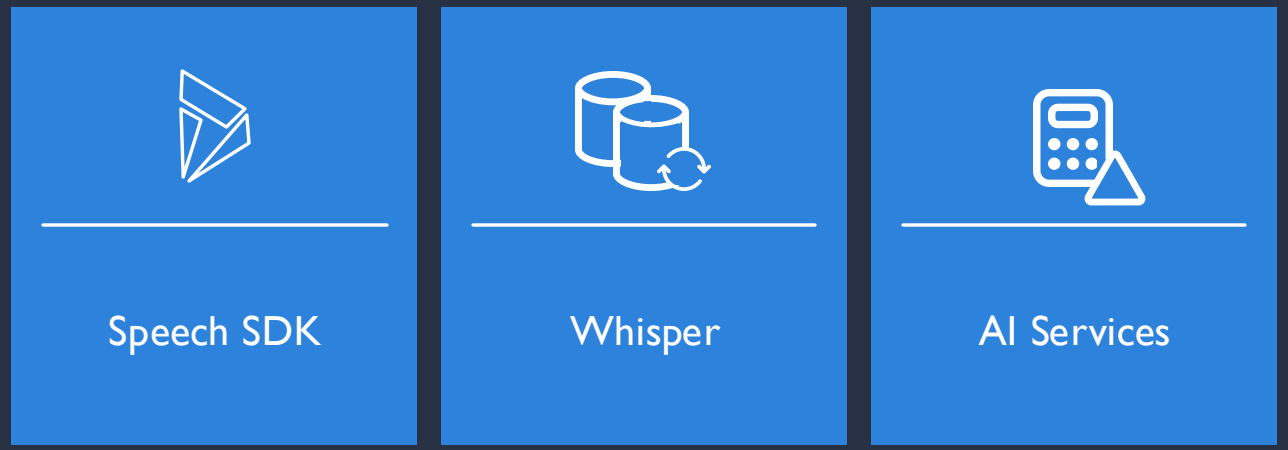
2017: **Google's Tacotron 2** model achieves high-quality speech synthesis, further pushing the boundaries of natural-sounding text-to-speech.

2020s: **Transformer-based models** like **VITS** (Variational Inference Text-to-Speech) improve the fluency and expressiveness of synthesized speech, used in applications such as virtual assistants and accessibility tools.



Resources you'll use

Azure provides a whole set of resources that you can use to build in Speech into your application



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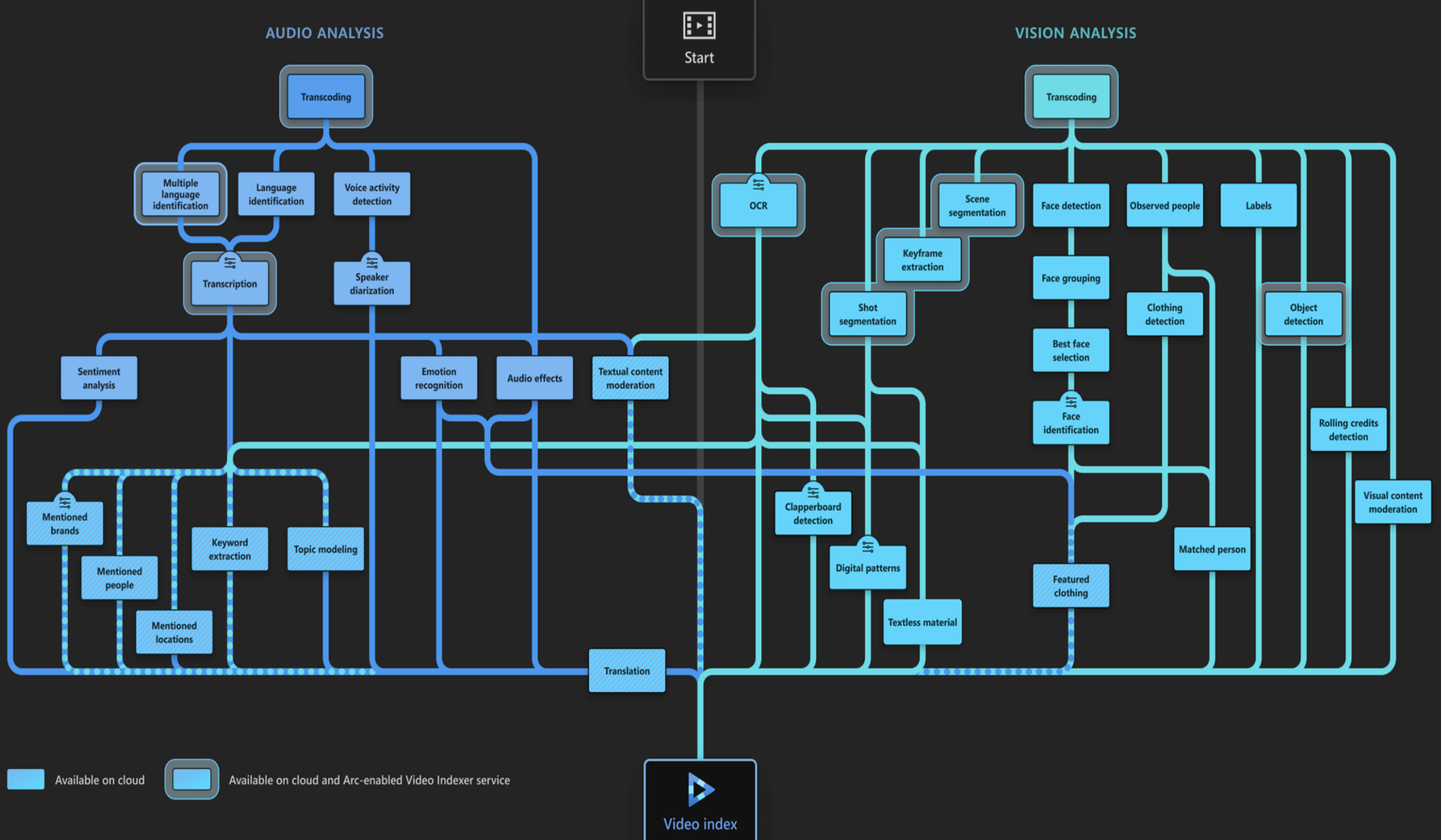
AUDIO ANALYSIS
Start
VISION ANALYSIS
Transcoding
Multiple language identification
Language identification
Voice activity detection
Transcription
Speaker diarization
Sentiment analysis
Emotion recognition
Audio effects
Textual content moderation
Mentioned brands
Mentioned people
Mentioned locations
Keyword extraction
Topic modeling
Translation
Transcoding
OCR
Scene segmentation
Keyframe extraction
Shot segmentation
Face detection
Face grouping
Best face selection
Face identification
Observed people
Clothing detection
Labels
Object detection
Rolling credits detection
Visual content moderation
Clapperboard detection
Digital patterns
Textless material
Featured clothing
Matched person
Available on cloud
Available on cloud and Arc-enabled Video Indexer service
Video index

Recording Audio

- Use **pyaudio** library to record audio
- Sampling up to 44100 Hz
- Write 128 bps across 2 channels
- Set quality using discrete scale
- Use standard python streams
- **Frames** are sampled and must appended to **stream**

```
audio_format = pyaudio.paInt16
encoder = lameenc.Encoder()
encoder.set_bit_rate(128)
encoder.set_in_sample_rate(44100)
encoder.set_channels(1)
encoder.set_quality(2) # 2-high 5-medium 7-low

p = pyaudio.PyAudio() # Create a PyAudio session

# Open the microphone stream
stream = p.open(format=audio_format,
channels=channels,rate=sample_rate,
input=True, frames_per_buffer=1024)
```

Encapsulating voice

- Use SSML to define voice
- Can contain content and characteristics
- Can contain many voices
- Contains different voice roles
- Define whether voice is happy, sad, angry, whispering etc.

```
<speak version="1.0" xmlns="http://www.w3.org/2001/10/synthesis"
xml:lang="en-US">
   <voice name="en-US-AvaMultilingualNeural">
      Good morning!
   </voice>
   <voice name="en-US-AndrewMultilingualNeural">
      Good morning to you too Ava!
   </voice>
</speak>
```

```
<speak version="1.0" xmlns="http://www.w3.org/2001/10/synthesis"
xml:lang="en-US">
 <voice name="en-US-AvaMultilingualNeural">
   <mstts:express-as role="YoungAdultFemale" style="calm/angry ...">
      Good morning!
   </mstts:express-as>
</speak>
```

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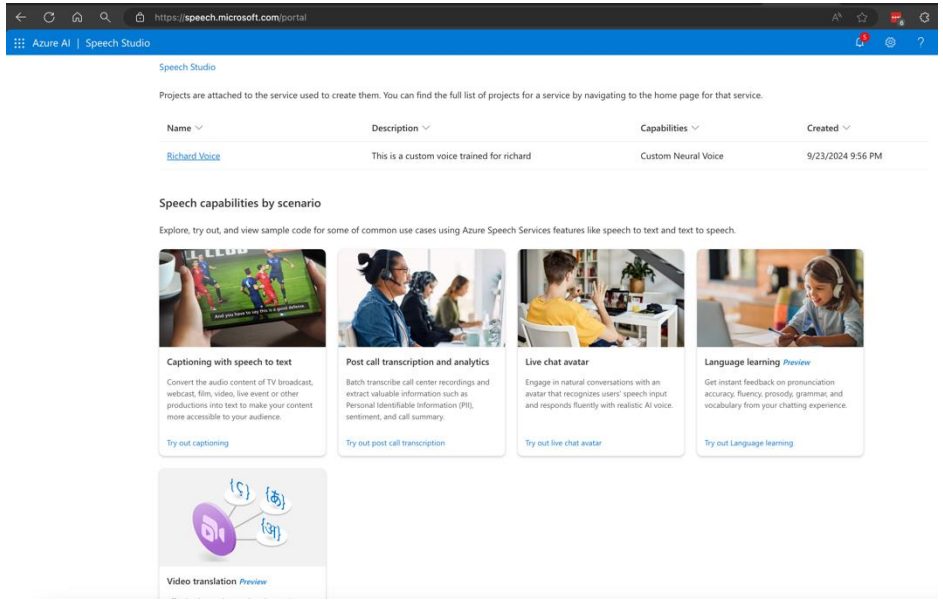
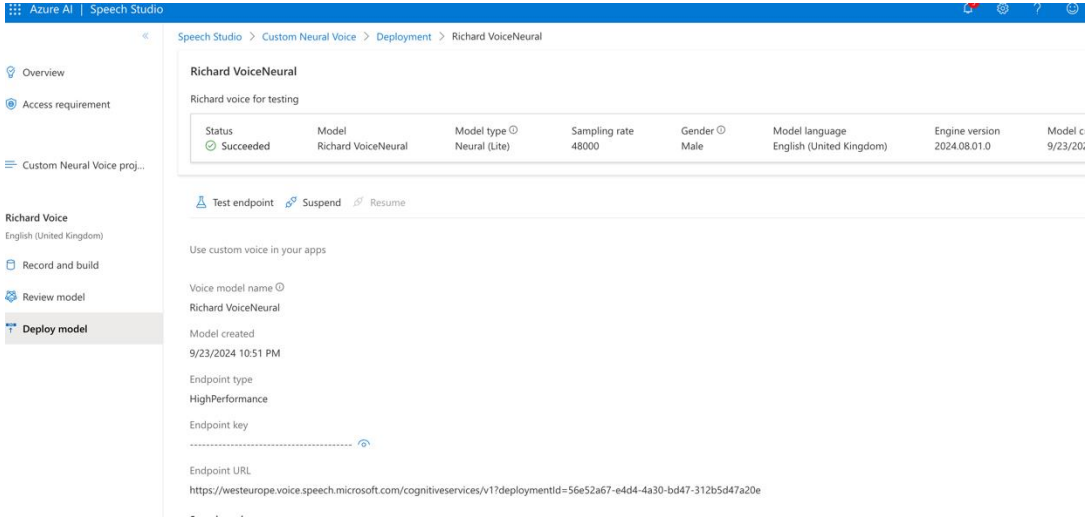
Using custom models

- Speech services allows you to train custom models
- Use models to provide the following:
  - Specialised vocab or domain specific terms for text to speech
  - Understand better accents and dialects (e.g. Scottish accent)
  - Cut out noise in noisier environments through better ”noisy” training set
  - Build in custom speech commands for security, home automation etc.
  - Speaker authentication and voice identification
  - Text to speech with custom voices
  - Multilingual support
  - Custom neural voice lite - Speech service - Azure AI services | Microsoft Learn

Speech Studio

- Contains a portal with different services
- Service include:
  - Transcription / language
  - Custom voices
  - Real-time or batch
  - Text to speech avatar
  - Custom voice commands





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Speech CLI

- Download and install spx
- Run voice tests from command line
- Customise voice using SSML
- Everything you can do through the API you can do with spx

```
> spx recognize –microphone
```

- spx recognize --file /path/to/file.wav
- spx synthesize --text "Testing synthesis using the Speech CLI" –speakers
- spx synthesize --text "Enjoy using the Speech CLI." --audio output my-sample.wav
- spx synthesize –voices
- spx synthesize --text "Bienvenue chez moi." --voice fr-FR-AlainNeural –speakers
- spx translate --microphone --source en-GB --target ru-RU

Enrolment

- For use cases where you need to build a voice database
- Voice database is secure and everyone has an id
- Has two APIs
  - Speaker verification
  - Speaker identification
- Useful if you want to identify people in a video or audio file where you have no context
- Samples at 16khz, single channel only
- Noise levels less than 2db

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DEMO: Recording Audio

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DEMO: Voice of Azure

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DEMO: Custom voices

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DEMO: Transcription

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Microsoft

DEMO: Video decomposition

ELASTACLOUD

Microsoft

DEMO: Video Indexing

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DEMO: Video and AI

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DEMO: Interview Mode

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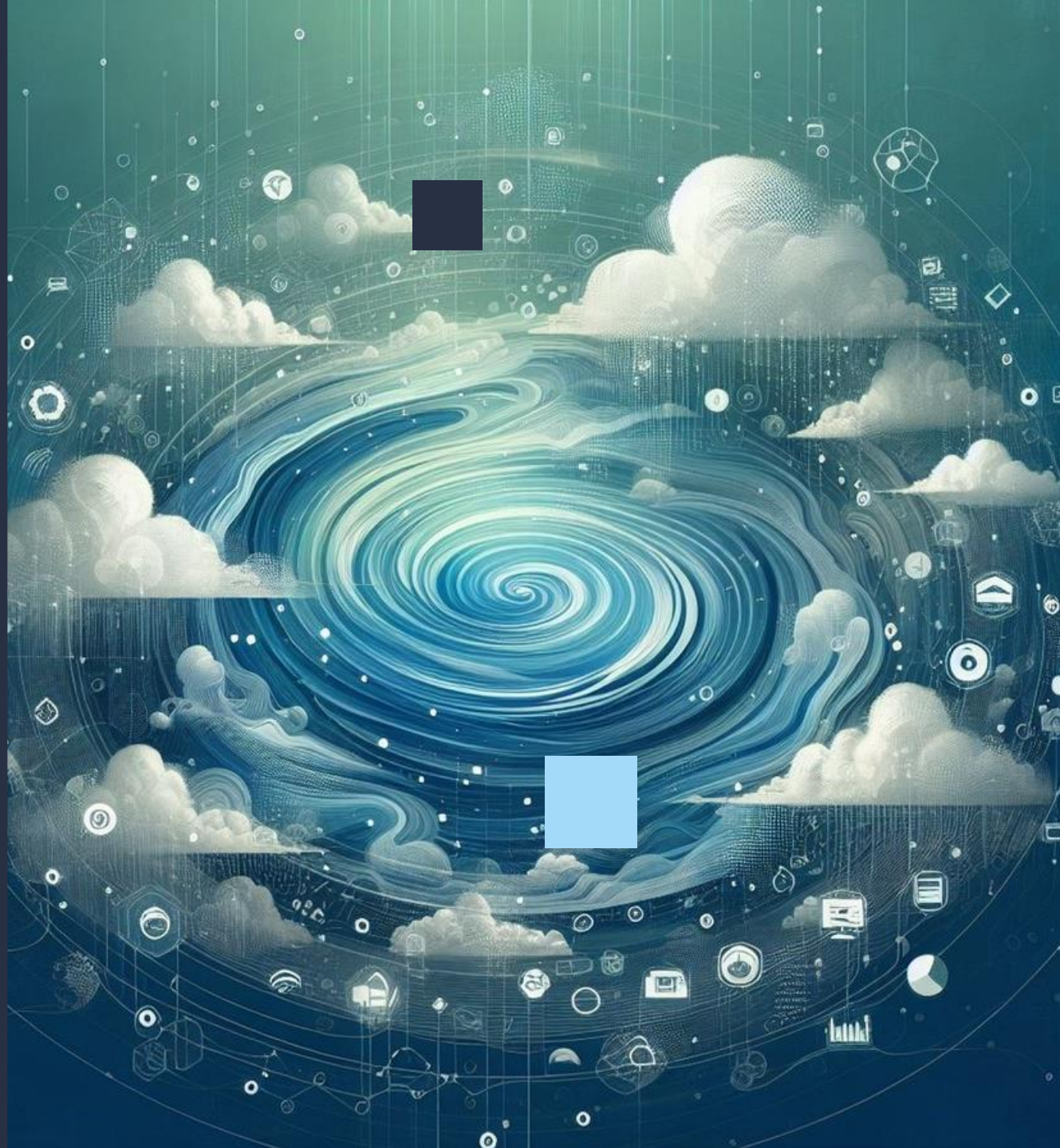

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