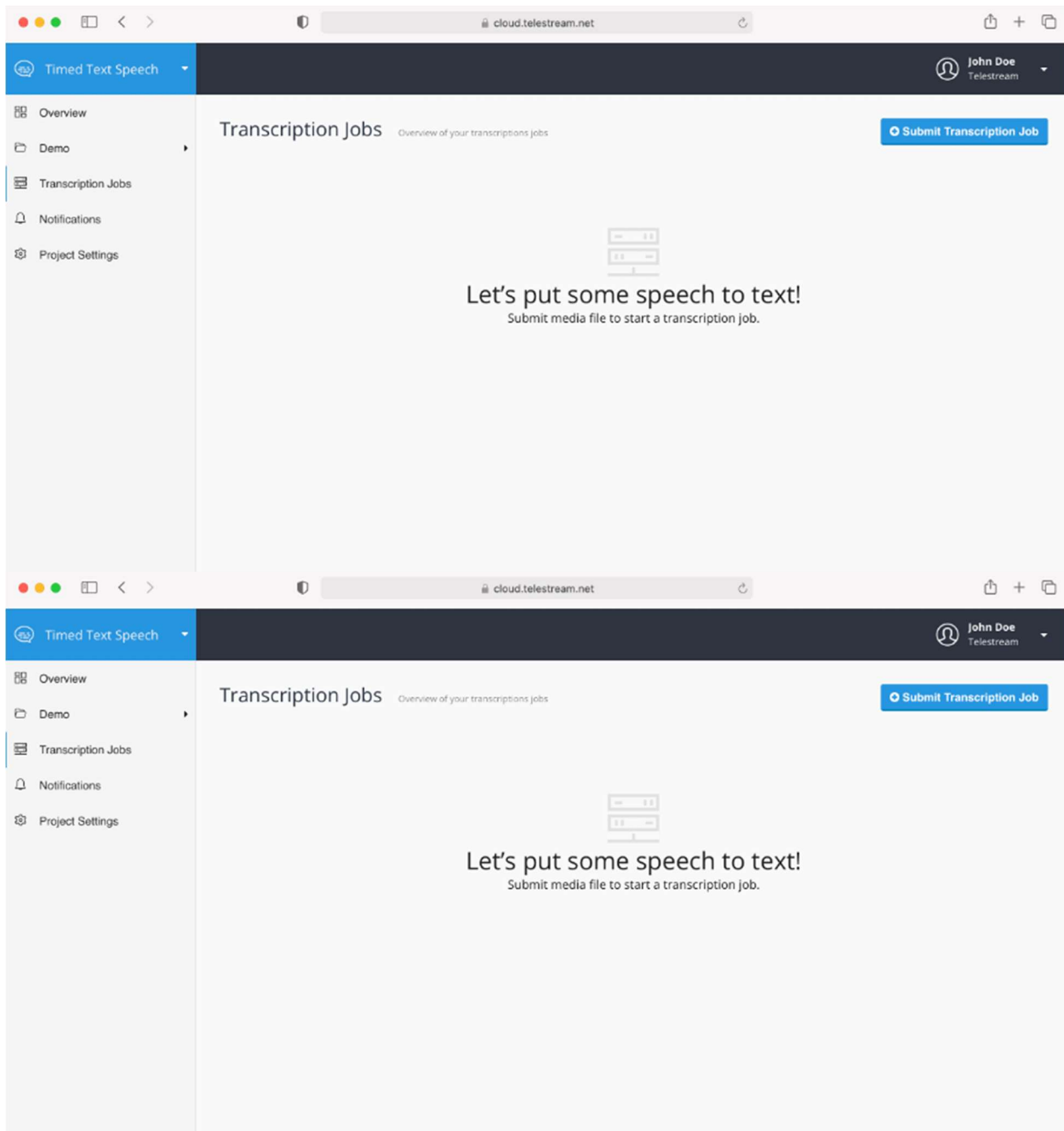


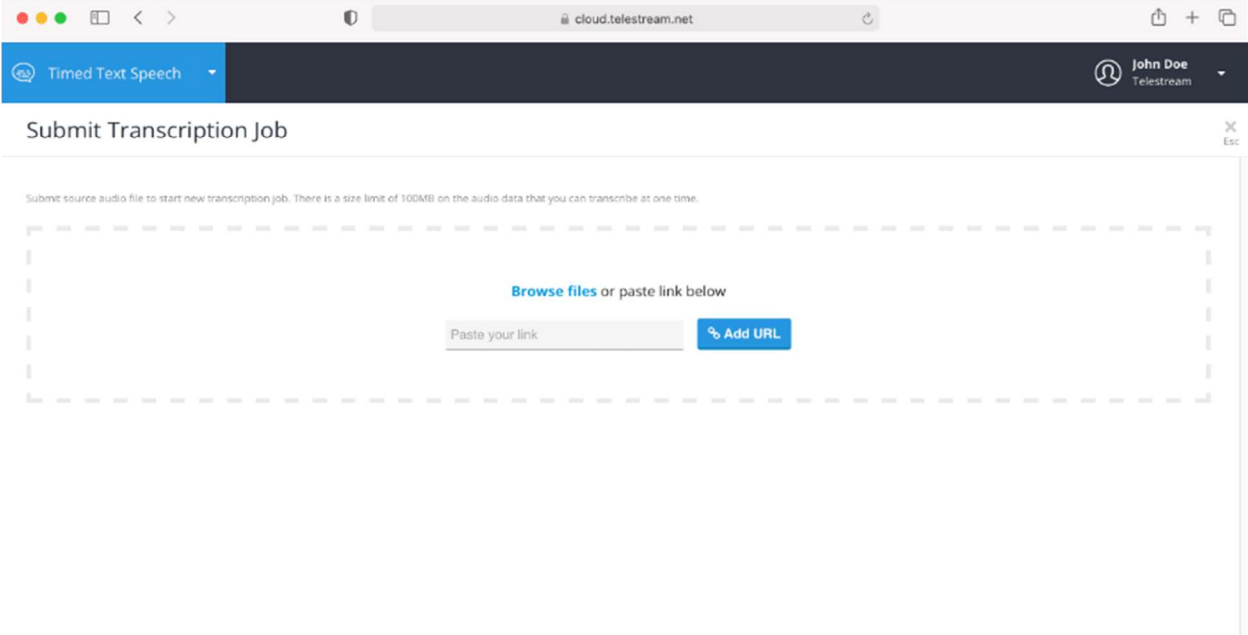
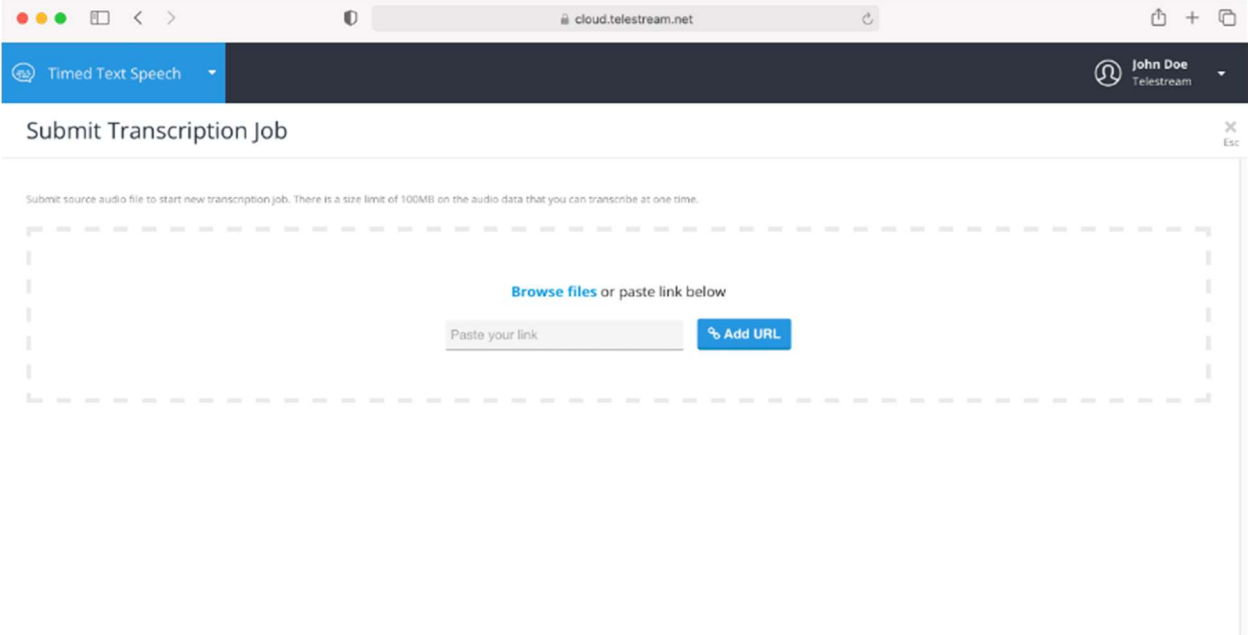
Running Transcription Jobs in the Cloud

Jobs can be submitted to Telestream Cloud Timed Text Speech in a few ways - API, from CaptionMaker and MacCaption, or in this case through web console.

Once logged in, select the Project that you'd like to use and you'll be looking at the Transcription Jobs list. This is where you can keep track of all jobs that have been processed or are currently in progress.



Click the Submit Transcription Job button to select files for processing. You can either drag & drop source files from your local disc or paste the URL to your media file. There is a size limit of 100MB on the audio data that you can transcribe at one time. If you upload a media file with both video and audio we will extract the audio track before the transcription process.



When ready, click the Submit Job button to start the upload and transcription process. You can follow the general progress in the jobs list. You can also click the

job in-progress from the list to see a more detailed view. When the transcription is finished it's time to move on to the final stage - review and editing.

Transcription Review and Editing

While Timed Text Speech transcription accuracy is usually very high you may want to review the final result anyway. That's why we added the ability to generate a proxy file (either audio or video) which makes the review process even easier.

- Overview
- Demo
- Transcription Jobs
- Notifications
- Project Settings

Job ID: b2bc8ef1323da2937462928d77ee62a1
[Back to Jobs List](#)

[Resubmit Job](#) [Delete](#)

Transcription Source

File URL: demo.mp4
File Size: 2526732
Audio Codec: UNK/UNK/16
Audio Bitrate: 256000
Audio Sample Rate: 16000

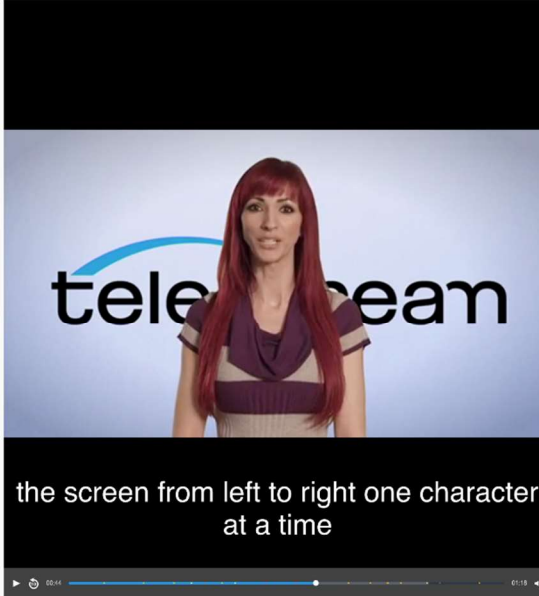
Transcription Files

Language: en-US
Duration: 00:01:18
Confidence: 85%
Profanity Filter: No

Progress 100%

Edit Transcription [Open in Editor](#)

[Review Again](#)



00:00:04.430 I am at the left of the screen.
00:00:05.940
00:00:06.230 so captions of what I say **block** the left of the screen too.
00:00:09.790
00:00:13.090 **no** **is** **at** **the** right of the screen so my captions appear at
00:00:16.540
00:00:16.540 the right
00:00:17.130
00:00:18.620 now **is** **off** **screen** to indicate that I'm off **screen** whenever
00:00:21.910
00:00:21.910 I say **is** **left** sized
00:00:23.570
00:00:24.380 now my name appears at the bottom of the screen **on** **off**
00:00:27.180
00:00:27.180 captions **what** I say at the top so **that** my name is not
00:00:30.080
00:00:30.080 covered by **captions**.
00:00:31.260
00:00:32.000 up until now we've been using **caption** on captions.
00:00:34.790
00:00:35.600 **when** **a** **new** **caption** **drops** **up** **remove** **the** **old** **caption** **disappears**.
00:00:38.940
00:00:40.580 this is a **new** **caption** **block** **is** **paired** **on**.
00:00:44.170
00:00:44.170 the screen from left to right one character at a time
00:00:47.800
00:00:48.700 now the role of mods
00:00:49.860
00:00:50.910 the mode is normally **live** for live TV programs.
00:00:53.450
00:00:53.840 caption **lines** **up** **one** **line** **at** **a** **time**.
00:00:56.280
00:00:57.200 captions **are** **scored** **at** **one** **time**.

The screenshot displays a web-based transcription interface. At the top, there's a navigation menu with options like 'Overview', 'Demo', 'Transcription Jobs', 'Notifications', and 'Project Settings'. The main area shows a job ID: b2bc8ef1323da2937462928d77ee62a1. Below this, there are sections for 'Transcription Source' and 'Transcription Files'. The 'Transcription Source' section lists details like File URL, File Size, Audio Codec, Audio Bitrate, and Audio Sample Rate. The 'Transcription Files' section shows Language, Duration, Confidence, and Profanity Filter. A progress bar indicates 100% completion. The main part of the interface is the 'Edit Transcription' section, which features a video player on the left and a transcription editor on the right. The video player shows a woman with red hair in front of a 'telestream' logo. The transcription editor shows a timeline with highlighted words and phrases in yellow, red, and orange, indicating areas of low confidence or alternative suggestions. The video player has a play button and a progress bar. The transcription editor has a 'Review Again' button.

The media file timeline has markings to show parts of the transcription which may need your special attention. This usually happens when the transcription model finds alternative words with similar confidence levels, or if the transcription confidence falls below a certain level.

The timeline is aligned with the transcription editor below, so any time you click on it you will be taken to the relevant place on the video timeline. Text highlighted in yellow, red, or orange means that the speech-to-text engine does not have high confidence that it located the correct word. This is where you need to step-in and either confirm it's the right word, correct it by choosing one of the available options, or enter the correct word manually. You can also edit whole lines at a time instead of single words.

There are also formatting options which allow you to split or merge lines to match the media file. All changes are visible on the fly so you have instant feedback on your actions. If at any time you need to review your changes there is history view available.

- Overview
- ⌵ Demo
- Transcription Jobs
- Notifications
- Project Settings


Job ID: b2bc8ef1323da2937462928d77ee62a1
[Back to Jobs List](#)

[Resume Job](#) [Delete](#)

Transcription Source	Transcription Files
File URL: demo.mp4 File Size: 252072 Audio Codec: L16/44100 Audio Bitrate: 756000 Audio Sample Rate: 15200	Language: en-US Duration: 00:01:18 Confidence: 85% Profanity Filter: No

Progress 100%

Edit Transcription [Approve Transcription](#) [Cancel](#)



00:00:04.430 I am at the left of the screen.
00:00:05.940
00:00:06.230 on captions of what I say **below** the left of the screen box.
00:00:09.790
00:00:13.690 **I** **am** the right of the screen so my captions appear at
00:00:16.540
00:00:16.540 the right
00:00:17.120
00:00:18.620 now **I** **am** off **screen** to indicate that I'm off **screen** whenever
00:00:21.910
00:00:21.910 I say **I** I talk sized
00:00:23.570
00:00:24.980 now my name appears at the bottom of the screen **my** **name**
00:00:27.180
00:00:27.180 captions what I say at the top so **my** my name is not
00:00:30.080
00:00:30.080 covered by **captions**
00:00:32.000 up until now we've been using **pop** on captions.
00:00:34.790
00:00:35.600 **pop** **pop** new caption pops up **pop** the old caption disappears
00:00:38.940
00:00:40.580 this is a **pop** **pop** **pop** one caption block is painted on
00:00:44.170
00:00:44.170 the screen from left to right one character at a time
00:00:47.600
00:00:48.700 now the role of music
00:00:49.880
00:00:50.910 this mode is normally **pop** for live TV programs
00:00:53.450
00:00:53.840 caption **pop** **pop** one line at a time
00:00:56.280
00:00:57.200 captions **pop** **pop** **pop** **pop**

00:01:01.600-00:01:02.010	line was approved within 2 minutes ago
00:00:13.690-00:00:13.730	line was approved less than a minute ago
00:01:01.700-00:01:04.073	Granule was changed to end 1 minute ago
00:01:00.780-00:01:03.090	SD was changed to size 2 minutes ago
00:00:24.910-00:00:24.940	pop was changed to pop up 5 minutes ago
00:00:23.590-00:00:24.070	popcom was changed to pop on 4 minutes ago
00:00:09.410-00:00:09.790	to was changed to too about 1 hour ago
00:01:03.280-00:01:03.950	case was changed to subcase about 1 hour ago

The screenshot displays a transcription software interface. At the top, there's a navigation menu with options like 'Overview', 'Demo', 'Transcription Jobs', 'Notifications', and 'Project Settings'. The main area shows a job ID: b2bc8ef1323da2937462928d77ee62a1. Below this, there are sections for 'Transcription Source' and 'Transcription Files'. The 'Transcription Source' section lists file details: File URL (demo.mpeg), File Size (2520752), Audio Codec (L16/48000), Audio Bitrate (256000), and Audio Sample Rate (16000). The 'Transcription Files' section shows Language (en-US), Duration (00:01:18), Confidence (85%), and Profanity Filter (No). A progress bar indicates 100% completion. Below this is the 'Edit Transcription' section, which features a video player on the left and a list of transcription corrections on the right. The video player shows a woman with red hair speaking in front of a 'telestream' logo. The correction list on the right includes various changes such as 'I am at the left of the screen', 'on captions of what I say', 'I say', 'the right of the screen so my captions appear at', 'the right', 'now I say off screen to indicate that I'm off screen whatever', 'I say I tall sized', 'now my name appears at the bottom of the screen', 'captions what I say at the top so my name is not covered by captions', 'up until now we've been using captions', 'new caption pops up', 'this is a new captions one caption block is painted on', 'the screen goes left to right one character at a time', 'now the role of male', 'this made a normally for live TV programs', 'caption has to go one line at a time', and 'captions can appear in'. Each correction includes a timestamp and a brief description of the change.

As soon as you're happy with the final result you can download the transcription file in one of the available formats - TXT, JSON, CSV or SRT and use it in your project.

Building Effective Custom Vocabulary

Speech-to-text engines are trained on a vast collection of sample recordings and texts. This means they perform well when your source is similar to "average" speech – i.e. typical conversations on common topics and using colloquial vocabulary and phrases that you would commonly find in your language. They will not perform as well if your source contains a lot of unique words or specialized terminology or phrases that the engine has not encountered before.

Using custom vocabulary allows you to inform the speech engine about unique words and phrases that are likely to occur in your source audio, so that it is more likely (but not guaranteed) to recognize them correctly.

Ideally, you should include things like proper nouns, acronyms, specialized terms, and short phrases which frequently occur in your audio but which are not part of typical every day conversations.

The downside of using custom vocabulary is that the words and phrases you include will be prioritized over more commonly spoken words and phrases. Using a large custom vocabulary with too many unnecessary words or phrases can actually decrease the accuracy of the results.

Following a few tips listed below will improve transcription accuracy.

Do Include

- Proper nouns (names of people, places, companies, etc.) – especially if they are from a different language, non-dictionary words, or words with an unusual spelling
- Acronyms (company names, abbreviations, etc.)
- Short phrases (less than 100 characters) which are unique to your source and are repeated often (e.g. a catch phrase often spoken by a character, or industry terminology)

Do not Include

- Words or phrases that are unlikely to occur in your source file
- Phrases longer than 100 characters
- Long phrases, sentences, or paragraphs that occur only once in your source
- Long lists of unrelated or unlikely words or phrases

Format

The custom vocabulary should be specified as a comma separated list of words or short phrases.

Longer phrases can be placed on separate lines (line delimited).

Limits

Please note these limits which are enforced by the API engine. Attempting to use a vocabulary file over the limits could result in the job failing.

These limits are so high that if you are approaching them, you should review the recommendations above.

- Number of phrases: 5000
- Characters per phrase: 100
- Total characters: 100,000

- **Getting the Best Results from Auto-Transcription**

- **Using Telestream Cloud for auto-transcription of media files is a great way to get started on a caption or subtitle project.**

- All submissions will result in an automatically generated transcript timed to match your media file. The timed transcript can be reviewed and edited on the fly in the Telestream Cloud console or directly populated into your MacCaption or CaptionMaker project. The more accurate the results the less clean up and editing is needed to make your transcript perfect. Below are some best practices and tips that can help increase the accuracy of the auto generated transcript from Timed Text Speech.

- **Isolating the Spoken Word**

For original content or media with multichannel audio, the dialogue-only track can be isolated to eliminate noise, music, and sound effects. This isolation can be done with any video editing software or audio production tool. By submitting audio with only spoken words to the Timed Text Speech engine, accuracy of the auto-generated transcripts can be greatly increased.

- To accomplish this a video editor can open their project in Adobe Premiere or Avid and silence all audio tracks which do not contain dialogue. Next, they can simply export an audio-only file. Timed Text Speech can handle audio files such as .mp3, aiff, and wav.

- In some cases your media file such a QuickTime or .MXF may contain multiple audio tracks. This could be a 5.1 mix, or isolated tracks for archival or transcoding purposes. Within MacCaption or CaptionMaker users have the option to select any of the audio tracks within the video file before submitting their project to Timed Text Speech. By default the software will submit tracks 1 and 2. If there is different audio track in the file which contains the spoken words to transcribe, the software can be configured to submit the alternate audio track instead. This means that the spoken-word-only track will be processed and the results will in turn be much more accurate.

- **Training the Speech Engine**

In many cases media files which require transcription may contain names, phrases, and acronyms that are not common. The speech engine may consistently get these wrong, causing users to manually correct the results again and again. To remedy this, Telestream Cloud's console offers a way

to train the Timed Text Speech engine by uploading a corpus text file, or by manually entering these uncommon words.

- This file is a simple plain text .txt document that contains a list of names and phrases that are used in a project. Users can upload this .txt document to any specific project that may require training to increase accuracy. We recommend that the .txt document contain a list of phrases on each line instead of only individual words.
- For example an **effective** corpus text document would like like this:
 - John Galveston
CEO of the Company
Working with CDN providers
Transcoding and captioning solutions
MacCaption Software
- An example of a corpus text file that is **not effective** looks like this:
 - John
Galveston
CDN
Providers
Transcoding
Captioning
MacCaption
Software
- By using phrases the speech engine knows what to expect and which other words are typically used with the new vocabulary. This added context means that results when using the custom vocabulary will greatly increase in accuracy.
- In some cases, creating a corpus text file for training is very easy and takes very little time. Some users simply repurpose the old transcripts or caption files from the same TV program or Project. For example, if a broadcaster needs to create a transcript for season 3 of a TV program, they can open the caption files from season 1 and 2 using MacCaption or CaptionMaker and export a corpus file which can be used for training Timed Text Speech. These 2 previous seasons contain the names and phrases that would greatly increase the vocabulary.
- Another way that users can leverage the vocabulary training of Timed Text Speech is when a rough transcript is already available of the media file prior to submission to Telestream Cloud. This rough transcript will also contain the names and key phrases for the project. Timed Text Speech would then automatically time the rough transcript and fill in the text that is missing.

- **Content Types Best Suited for Automatic Speech Recognition (ASR)**
The type of video content plays an important role in the level of accuracy achieved using auto-transcription software. For example, a news show with a professional announcer and clear studio audio will have great accuracy compared to a video shot outdoors in a noisy environment on a mobile phone. In addition, loud music, singing, and shouting will also bring down the level of accuracy. There are also cases where speakers may change their voice to provide dialogue for children's programming or for dramatic effect. This means that a speech engine that is designed and trained for standard voices may not be able to understand these voice tones. Generally speaking, projects with clear studio quality recordings, minimal background noise, and a professional speaker will always result in the best accuracy.
- **Creating a Proxy**
Professional video companies generally work with high quality video master files called mezzanine files. These files are used the same way tape masters were used in the old days. The original video must be uncompressed or high bitrate when submitted for processing. This is not the case for Timed Text Speech workflows. Because Telestream Cloud requires only the audio content for auto-transcription, users can submit a low bitrate MP4, or just the audio file. As long as the audio quality is good you will achieve high quality results, the video quality or resolution does not affect your results.
- **Voiceover for the Purpose of ASR**
For video editing workflows, it's quite common for editors to do a rough voiceover when editing prior to bringing in voice talent to the studio to record the final audio. This also provides an opportunity for video editors to re-speak any portions of the video project that do not have clear audio. This rough voice over can then be exported from the video editing system and submitted to Timed Text Speech for processing. This means that results will have a greater accuracy than the original audio.