# SURT 2.0: Advances in Transducer-based Multi-talker Speech Recognition 

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## Motivation

- Existing ASR systems are mostly geared towards single-user applications.
- We want to build systems that answer "who spoke what" for free-flowing multi-party conversations, in real-time
- How to train efficient end-to-end neural models for this task?



## Challenges

Multi-talker conversations contain overlapping speech and back-channels.

There is limited amount of real data available for training end-to-end neural models.

Training such systems requires large
Compute computational resources

Continuous Streaming Multi-talker ASR


| Continuous |
| ---: |
| - No need of external <br> segmentation |


| Streaming |
| :---: |
| -Overlapping speakers <br> transcribed simultaneously |

## Streaming Unmixing and Recognition Transducer

## Shorter mixtures with more turn-taking

Zipformer for efficient sequence modeling


Results

- Experiments on meeting corpora: LibriCSS, AMI, ICSI
- LibriCSS is "simulated"; AMI and ICSI are real meetings
- SURT 2.0 obtains $44.6 \%$ and $32.2 \%$ WER on real far-field meetings.



## Analysis

1. Single speaker pre-training is critical.

2. Auxiliary objectives improve performance on high-overlap conditions.


## References

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