様　式　Ｃ－１９、Ｆ－１９－１、Ｚ－１９　（共通）

科学研究費助成事業　　研究成果報告書

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研究課題名（和文）Phantom in the Opera: the Vulnerabilities of Speech Interface for Robotic Dialogue System

研究課題名（英文）Phantom in the Opera: the Vulnerabilities of Speech Interface for Robotic Dialogue System

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研究成果の概要（和文）：このプロジェクトでは、音声認識システムの原理を研究し、攻撃の詳細を調査。調査結果をまとめ、音声認識システムの改善方法を提案しました。研究範囲を普遍的視点で拡大し、音声関連システムにも同様の攻撃が共存できることを示しました。敵対的攻撃をノイズとみなし、音声強化・モデリング・ポストプロセッシング法を組み合わせて対処。InterspeechやICASSPなどが成果を認め、2冊の本に紹介され、関西国立図書館に収められました。これはAIシステムの安全性と信頼性の確保に貢献しています。

研究成果の学術的意義や社会的意義
The development of deep neural networks has been progressing rapidly and the evolution of speech recognition systems has been incredibly fast. The study aims to provide researchers with ideas on improving system security in light of the increasingly severe security issues.

研究成果の概要（英文）：In this project, we carefully studied the principles of speech recognition systems and researched all possible attack details. We summarized our findings in a review and proposed methods for improving the front-end and back-end of speech recognition systems.
We expanded our research scope with a universal point of view. Similar attacks can co-exist in speech-related systems, not just speech recognition systems. We also consider adversarial attacks as particular noise, then combining traditional speech enhancement, modeling, and post-processing methods in system development can sufficiently deal with this attack.
Top journals and conferences in the speech field accepted our achievements, such as Interspeech and ICASSP. Above two years of research achievement have been introduced into two books (ISBN: 978-4-904020-26-5, ISBN: 978-4-904020-28-9) by NICT and stored in the national library Kansai. These efforts are our contribution to ensuring the security and reliability of AI systems.

研究分野：知覚情報処理

キーワード：speech recognition　adversarial attack　privacy perserving　deepfake detection　spoken dialogue　federated learning　security

１．研究開始当初の背景

Nowadays, spoken dialogue systems have been widely used in our society. The automatic speech recognition (ASR) module, as these dialogue systems' most natural human-machine interface, is based on deep neural networks (DNNs). The development of deep neural networks has been progressing rapidly, from basic DNNs to end-to-end models, to the popular self-supervised learning models in recent years, and now to the revolutionary large language models (LLMs). However, DNN is known to be vulnerable to adversarial examples (or attacks). This is a severe problem.

２．研究の目的

The study of adversarial attacks on DNN-based speech recognition systems serves as a starting point for our research, aiming to provide speech development researchers with ideas on improving system security in light of the increasingly severe security issues related to speech-based systems.

３．研究の方法

Current academia is focusing on designing individual purposed perfect adversarial attacks on Automatic Speech Recognition (ASR) systems while still sounding normal to human listeners with the following focuses:

1. Black-box attacks: In black-box attacks, the attacker has limited or no access to the internals of the target model, such as its architecture or weights. They craft adversarial examples by exploiting the transferability of adversarial perturbations across models or by querying the target model and using its output to estimate its behavior.

2. Untargeted attacks: Adversarial attacks on speech systems can be categorized into targeted and untargeted attacks. Targeted attacks aim to make the system produce a specific, desired output, while untargeted attacks cause incorrect output without a specific target.

3. Real-world applicability: Researchers are increasingly focusing on adversarial attacks that can be effective in real-world settings, such as over-the-air attacks or attacks robust to various environmental conditions like background noise.

These works studying individual threats are very important contributions to the community. However, **other substantial threats co-exist in real-world scenarios and adversarial attacks**. These attacks can compromise the integrity of machine learning models and seriously affect critical applications, such as autonomous vehicles, healthcare, and security systems.

In our proposed method, we propose **a universal point of view on dealing with these threats**.

1. **Defense of the whole system pipeline and taking adversarial audio as noise**: The speech recognition module is not isolated in the industry system pipeline. Taking adversarial audio as a type of noise, we confirm that three traditional methods can effectively deal with adversarial attacks: 1. Changing/detecting front-end input signal (denoising, adding noise, or detect-then-reject), 2. Changing models (tuning structure or retraining), and 3. Correct by back-end language models. All these methods exist in current speech systems.

2. **Extend to concerning the robustness of other speech-related models**: Similar to the adversarial attack, other substantial threats exist. **(1) Deepfakes** are manipulated media, particularly images and videos, created using advanced

machine-learning techniques. Deepfakes can spread misinformation, manipulate public opinion, and harm reputations, posing a severe threat to individuals, organizations, and society. **(2) Privacy leakage** refers to unauthorized access or disclosure of sensitive personal information, often resulting from inadequate data protection measures or malicious activities. Privacy leakage can lead to identity theft, financial loss, and damage to an individual's reputation. **(3) Adversarial attacks to other speech models:** the same immediate impact as deepfakes or privacy leakage, adversarial attacks on speaker recognition systems can still be a significant threat.

４．研究成果

In the first year of this project, we carefully studied the principles of speech recognition systems and researched all possible attack details. We summarized our findings in a review and proposed methods for improving the front-end and back-end of speech recognition systems. Our achievements were accepted by top conferences in the speech field, such as Interspeech and ICASSP, as followings:

- We construct speech recognition systems with recent popular training toolkits and neural network types (accepted in Journals and conferences, e.g., ICASSP2022).
- We did surveys for the current attack methods (accepted in Journal).
- We implement robust adversarial attacks using the Kaldi-based ASR systems (accepted in SLT2020).
- We are also happy to see that this framework can be used to protect sensitive speech content (accepted in LREC 2022).
- To defend against attacks, we find that adversarial audios are very sensitive. Moreover, the feature of its spectrogram is very different from the human voice, and it can be treated as a special kind of noise. We constructed speech enhancement systems and studied their mechanism this year (accepted in Journals and conferences, e.g., ICASSP2022).

In the project's second year, we discovered that attacks on speech recognition systems are not only limited to the content of speech recognition but also involve more attacks related to speaker attributes. Therefore, we expanded our research scope with a universal point of view.

- We continued research on the front-end method about synthesized speech quality estimation, signal-to-noise ratio estimation framework (accepted in conferences, such as APSIPA2022, EUSIPCO2022, and Interspeech2022), and back-end of the language model, speech recognition/translation systems and other down-streaming systems (accepted in Journals and conferences, e.g., JNLP, International Journal of Asian Language Processing, Interspeech2022, ICASSP2023).
- We participated in several privacy attack competitions, proposing our deepfake detection (accepted by Interspeech2022). Moreover, we published our first database following security protection law (O-COCOSDA2022).
- At the same time, we expanded our research from speech recognition systems to spoken dialogue systems (accepted in Sigdial2022).
- We also studied the state-of-the-art modeling training method, self-supervised training (accepted in Interspeech2022), and federated learning (accepted in ICASSP2023).

Above two years of research achievement have been introduced into two books (ISBN: 978-4-904020-26-5, ISBN: 978-4-904020-28-9) by NICT and stored in the national library Kansai.

These efforts have laid the foundation for further research to ensure the security and reliability of AI systems.

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３．雑誌名 Journal of Natural Language Processing	６．最初と最後の頁 611～637
掲載論文のDOI（デジタルオブジェクト識別子） 10.5715/jnlp.29.611	査読の有無 有
オープンアクセス オープンアクセスとしている（また、その予定である）	国際共著 －

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掲載論文のDOI（デジタルオブジェクト識別子） 10.1142/S2717554522500072	査読の有無 有
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3．書名 Phantom in the Opera: The Vulnerabilities of Speech-based Artificial Intelligence Systems	

〔産業財産権〕

〔その他〕

The international collaboration with China is based on NICT's international collaboration activity in 2021. In 2022, we turn to Singapore according to NICT's rule.

情報通信研究機構の研究成果として、各年ごとの発表論文を日付順で紹介します。
https://www.nict.go.jp/outcome/journals/journals_2021_j.html
情報通信研究機構の研究成果として、各年ごとの発表論文を日付順で紹介します。
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Lab homepage of Sheng Li
https://ast-astrec.nict.go.jp/member/sheng-li/index.html

6．研究組織

	氏名 （ローマ字氏名） （研究者番号）	所属研究機関・部局・職 （機関番号）	備考

7．科研費を使用して開催した国際研究集会

〔国際研究集会〕　計0件

8．本研究に関連して実施した国際共同研究の実施状況

共同研究相手国	相手方研究機関

中国	Tianjin University	Xinjiang University	Royal Flush AI Research Inc.	
シンガポール	Nanyang Technological University			