An assessment of automatic speaker verification vulnerabilities to replay spoofing attacks

Artur Janicki, Federico Alegre and Nicholas Evans

Abstract—This article analyses the threat of replay spoofing or presentation attacks in the context of automatic speaker verification (ASV). Replay may present a significant threat to their reliability. As relatively high-technology attacks, speech synthesis and voice conversion, which have thus far received far greater attention in the literature, are probably beyond the means of the average fraudster. The implementation of replay attacks, in contrast, requires no specific expertise nor sophisticated equipment. Replay attacks are thus likely to be the most prolific in practice. This paper compares the threat of replay attacks to that of speech synthesis and voice conversion. The comparison is performed using strictly controlled protocols and with seven different ASV systems including a state-of-the-art iVector system with probabilistic linear discriminant analysis. While comparatively higher-effort spoofing attacks have received far greater attention in the recent past, experiments show, perhaps unsurprisingly, that low-effort replay attacks represent a greater threat, currently under-researched. The paper also describes and assesses two replay attack countermeasures. A relatively new approach based on the local binary pattern (LBP) analysis of speech spectrograms is shown to outperform a competing approach based on the detection of far-field recordings.

Index Terms—speaker verification, spoofing, replay, countermeasures, local binary patterns.

I. INTRODUCTION

Spoofing refers to the presentation of a falsified or manipulated sample to the sensor of a biometric system in order to provoke a high score and thus illegitimate verification. In recent years the automatic speaker verification (ASV) community has started to investigate spoofing and countermeasures actively [1], [2]. A growing body of independent work has demonstrated the vulnerability of ASV systems to spoofing through replayed speech [3], [4], impersonation [5], [6], voice conversion [7], [8], speech synthesis [9], [10] and attacks with non-speech, artificial, tone-like signals [11], [12].

Common to the bulk of previous work is a focus on attacks which require either specific expertise, e.g. impersonation, or high-level technology, e.g. speech synthesis and voice conversion. Only replay attacks can be performed with ease, requiring neither specialist expertise nor sophisticated equipment. They are easily implemented with discreet, high-quality

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audio equipment, now available to the masses. Accordingly, it is reasonable to assume that replay attacks represent a tangible threat and that they will be the most prolific in practice.

Only few studies have addressed replay. The work in [3] assessed the vulnerabilities of a hidden Markov model (HMM) based, text-dependent ASV system using concatenated digits. While results showed that replay attacks are highly effective, experiments were conducted with data collected from only two speakers. The work in [4] investigated replay using recordings which were collected with close-talk or far-field microphones and then replayed over an analogue or digital telephony channel. The work was conducted with data collected from five speakers and demonstrated the vulnerability of a joint factor analysis (JFA) ASV system; the false acceptance rate (FAR) at the equal error rate (EER) threshold increased from 1% to almost 70%. The authors in [14] investigated a text-dependent ASV system exposed to speech replayed using a laptop computer. Using the large, standard and publicly available RSR2015 corpus, this work showed that the EER for an HMM system increased from approximately 4% to more than 20%.

Missing from the literature, however, is a reliable comparative assessment of replay attacks to speech synthesis and voice conversion using large, standard databases and using a representative range of replay scenarios. Such a study is needed in order to help prioritise future work to develop countermeasures for the protection of ASV from spoofing. This paper accordingly aims to assess ASV vulnerabilities to replay attacks using the same ASV systems and corpora used in previous assessments involving speech synthesis and voice conversion spoofing attacks. In addition, the paper investigates the effectiveness of new countermeasures which aim to distinguish between genuine and replayed speech.

The paper is organised as follows. Section 2 describes speech synthesis and voice conversion spoofing attacks with a comparison to replay attacks. Section 3 presents previous and ongoing work to develop countermeasures against replay attacks, including the authors' work using the local binary pattern analysis of speech spectrograms. A common experimental framework for the assessment of both vulnerabilities and countermeasures is presented in Section 4. Results are presented in Section 5 and our conclusions and ideas for future work are presented in Section 6.

II. SPOOFING SPEAKER VERIFICATION SYSTEMS

This section describes speech synthesis, voice conversion and replay spoofing attacks and specific implementations. In

general, a spoofed speech signal $s(t)$ is generated from the speech signal of a target speaker $x(t)$. Whereas the input to a speech synthesis system is a text string, that to voice conversion and replay attacks originates from a third speech signal of a source speaker (spoofer) $y(t)$.

A. Speech synthesis

There is a large variety of speech synthesis algorithms, such as formant, diphone, unit-selection and statistical parametric based approaches, in addition to more recent deep-neural network architectures. Whatever the approach, the aim is to generate intelligible, natural speech for a given text string c. In the context of spoofing, a synthetic speech signal is generated according to:

$$s(t) = g_{x(t)}(c), \tag{1}$$

where $g_{x(t)}$ denotes a text-to-speech mapping generated by a synthesis system with speech units or acoustic models extracted or learned from the speech signal of a target speaker $x(t)$. While unit-selection approaches generally require large amounts of speaker-specific data to learn the mapping function $g_{x(t)}$, statistical parametric approaches can synthesize convincing speech signals with the adaptation of well-trained models using relatively small quantities of speaker-specific data.

Our approach to statistical parametric speech synthesis uses hidden Markov models following the approach described in [15]. The specific implementation uses the HMM-based Speech Synthesis System (HTS)[1] where speech signals are parametrised by STRAIGHT (Speech Transformation and Representation using Adaptive Interpolation of weiGHTed spectrum) features, Mel-cepstrum coefficients and the logarithm of the fundamental frequency ($\log F_0$) with their delta and acceleration coefficients. Acoustic spectral characteristics and duration probabilities are modelled using multispace distribution hidden semi-Markov models (MSD-HSMM) [16]. Speaker dependent excitation, spectral and duration models are adapted from corresponding independent models according to a speaker adaptation strategy referred to as constrained structural maximum a posteriori linear regression (CSMAPLR) [17]. Finally, time domain signals are synthesised using a vocoder based on Mel-logarithmic spectrum approximation (MLSA) filters. They correspond to STRAIGHT Mel-cepstral coefficients and are driven by a mixed excitation signal and waveforms reconstructed using the pitch synchronous overlap add (PSOLA) method.

B. Voice conversion

Voice conversion has been used to explore ASV spoofing since the late 90s [8]. One successful approaches involves so-called Gaussian-dependent filtering [18]. Here, the spoofing signal $s(t)$ (or $S(f)$ in the spectral domain) is generated by filtering at the frame level the speech signal of the source or spoofer $y(t)$. In the spectral domain it can be represented as follows:

$$S(f) = \frac{|H_x(f)|}{|H_y(f)|} Y(f) \tag{2}$$

where $H_x(f)$ and $H_y(f)$ are the vocal tract transfer functions of the targeted speaker and the spoofer respectively. $Y(f)$ is the spoofer's speech signal in the spectral domain whereas $S(f)$ denotes the result after voice conversion. A time-domain, converted signal is recovered as follows:

$$s(t) = IFFT(\frac{|H_x(f)|}{|H_y(f)|}) * y(t) \tag{3}$$

where $*$ denotes convolution. As such, $y(t)$ is mapped or converted towards the target in a spectral-envelope sense, and is sufficient to overcome most ASV systems **??**.

$H_x(f)$ is determined from a set of two Gaussian mixture models (GMMs). The first, denoted as the automatic speaker recognition (asr) model in the original work, is related to ASV feature space and utilised for the calculation of a posteriori probabilities. The second, denoted as the filtering (fil) model, is a tied model of linear predictive cepstral coding (LPCC) coefficients from which $H_x(f)$ is derived. LPCC filter parameters are obtained according to:

$$x_{fil} = \sum_{i=1}^{M} p(g_{asr}^i|y_{asr})\mu_{fil}^i \tag{4}$$

where $p(g_{asr}^i|y_{asr})$ is the a posteriori probability of Gaussian component g_{asr}^i given the frame y_{asr} and μ_{fil}^i is the mean of component g_{fil}^i which is tied to g_{asr}^i. $H_x(f)$ is estimated from x_{fil} using an LPCC-to-LPC transformation and a time-domain signal is synthesised from converted frames with a standard overlap-add technique. Full details can be found in [18], [19], [20].

C. Replay

Replay attacks are an example of low-effort spoofing; they require simply the replaying of a previously captured speech signal. In the absence of suitable countermeasures and considering the widespread availability of consumer devices with high-quality sound systems, replay attacks can typically be realised with ease. Furthermore, used either directly, or through the cutting and pasting of short speech intervals, replayed speech has potential to overcome both text-dependent and text-independent ASV systems. Even though the processes of recording and replaying introduce additive acoustic and convolutive channel and transducer noise, these effects can be attenuated by noise and other intersession (channel) variability compensation techniques common to most modern recognition systems. These factors point towards the tangible threat posed by replay attacks.

Ignoring ambient noise in the acoustic environment (which is in any case not specific to the replay spoofing scenario), replayed speech can be represented as:

$$s(t) = x(t) * h(t), \tag{5}$$

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[1]http://hts.sp.nitech.ac.jp/

Attack	Naïve impostor	Replay	Voice conversion	Speech synthesis
Input	impostor speech	target speech	impostor speech	text
Effort	zero	low	medium-high	high
Effectiveness	low	(?)	medium-high	high

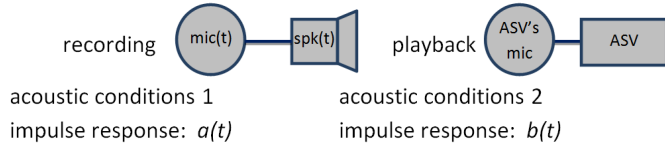TABLE I: Comparison of four different attacks in terms of speech used, required effort and effectiveness.

Fig. 1: A schematic diagram of the assumed replay attack configuration.

where * denotes convolution. The composite replay effects denoted by $h(t)$ include the impulse responses of replay hardware and the replay environment. It is composed by:

$$h(t) = mic(t) * a(t) * spk(t) * b(t) \qquad (6)$$

where $mic(t)$ and $spk(t)$ are impulse responses of the microphone and the speaker, respectively, and where $a(t)$ and $b(t)$ are the respective impulse responses of the recording and replay environments. This scenario is illustrated in Fig. 1. Replay attacks may thus be emulated through the convolution of a source signal $s(t)$ with typical impulse responses representing the different replay components.

D. Qualitative comparison

Replay, voice conversion and speech synthesis spoofing are forms of *concerted-effort* impostor attacks, as opposed to the naïve or *zero-effort* impostor attacks normally used to assess ASV system performance. A qualitative comparison of all four is illustrated in Table I, ordered by the level of effort or expertise needed to implement each attack successfully [2].

Compared to naïve impostor attacks, replay attacks require slightly increased effort; they require recording and replaying. Voice conversion and speech synthesis attacks require specialised, often complex algorithms, in addition to any recording hardware to collect, analyse and parametrise the target and any other auxiliary speech data. They belong to a class of higher-effort spoofing attacks. While voice conversion is based upon the conversion of one speech signal to another, speech synthesis converts a text string to a speech signal, which requires a comparatively higher level of effort or expertise.

One may reasonably suppose that the effectiveness of each attack is correlated with the effort involved in their implementation; the higher the effort, the more effective the algorithm and hence the greater the impact on ASV performance. Replay attacks are then assumed to pose only a low threat. However, the proof of concept presented in [21] suggests the contrary, showing that replay attacks pose a significant threat, being effective in overcoming an ASV system while being the easiest implemented spoofing attack. It is the objective of this paper

to investigate these contradictory findings and to compare objectively and quantitatively the comparative threat of replay spoofing to those of speech synthesis and voice conversion.

III. REPLAY COUNTERMEASURES

Attention now turns to the *detection* of replay spoofing attacks. Given that only little work has investigated ASV vulnerabilities to such attacks, it is hardly surprising that work to develop anti-spoofing countermeasures is similarly limited. This section briefly reviews that past work and then describes two particular replay countermeasures which are explored further in this paper.

A. General approaches

One obvious approach to replay detection involves challenge-response systems which require the speaker to utter a prompted phrase [22]. Challenge-response mechanisms are a form of passive countermeasure. While having potential in preventing some forms of replay attack for some ASV systems, challenge-response countermeasures are not without negative impacts on usability which may render them undesirable for other ASV systems.

Active countermeasures have also been proposed. One such approach involves the storing of previous access attempts and their comparison to new attempts [23]. New access attempts which are deemed too close to previous attempts are rejected. A somewhat similar technique is proposed in [14], where the authors compare spectral bitmaps between access trials and previously stored recordings in a text-dependent ASV scenario.

Other, more generally applicable methods not restricted to any particular ASV scenario are based on the detection of unexpected channel artefacts indicative of recording and replaying. Two such algorithms were reported in [24] for which the EER for a baseline GMM-UBM system was shown to decrease from 40% to 10% with active countermeasures. Channel detection is the basis of the first approach investigated further in this paper.

B. Far-field channel detection

Many scenarios in which user authentication is performed by ASV involve so-called close-talk speech, i.e. situations where speech is collected from an in-situ or closely positioned microphone. Examples include telephone and logical access scenarios or critical infrastructure protection and physical access scenarios. In contrast, since they are likely to be collected surreptitiously or at-distance, replay recordings will exhibit far-field channel effects, effects which can be measured and consequently used to detect replayed speech.

This idea was first investigated in [25]. The work compared close-talk and far-field speech signals parametrised according to 12 channel-sensitive features:

- spectral ratio – sub-band energy ratio from 0-2 kHz and from 2-4 kHz;
- low frequency ratio – sub-band energy ratio from 100-300 Hz and from 300-500 Hz, calculated using speech frames only;
- modulation index, and
- nine sub-band modulation indices – see [25] for precise sub-band bandwidths.

The spectral ratio reflect reflects the level of spectrum flattening or noise and reverberation introduced by far-field recording. The low frequency ratio reflects the level of high-pass filtering, an artefact typical of speech signals produced by small loudspeakers. The total and sub-band modulation indices reflect the level of additive and, specifically, coloured noise; higher levels of noise present in replay recordings result in lower than average modulation indices. Experiments run with the support vector machine (SVM) as a classifier showed that far-field recordings could be detected with 90% accuracy.

C. Local binary patterns

The approach to replay detection proposed in [21] is based on the local binary pattern (LBP) analysis of speech spectrograms. Inspired by the original application to image texture analysis [26], the idea was introduced as an ASV spoofing countermeasure in [27]. As illustrated in Fig. 2, LBP analysis is applied to a mel-scaled cepstrogram with appended dynamic features. Modifications made through spoofing are assumed to disturb the natural 'texture' of genuine speech, attributes which are readily detected with LBP.

The standard LBP operator is a non-parametric 3x3 kernel which assigns a binary code to each pixel in an image according to the comparison of its intensity value to that of its eight surrounding pixels [26]. A binary value of 1 is assigned when the intensity of neighbouring pixels (here feature components) is higher, whereas a value of 0 is assigned when neighbouring pixels are of lower or equal intensity. Each pixel is thus assigned one of $2^8 = 256$ binary patterns.

LBPs are determined for each pixel in the mel-scaled cepstrogram thus resulting in a new matrix of reduced dynamic range, here referred to as a *textrogram*. The textrogram captures short-time feature motion beyond that in conventional dynamic parametrisations. Normalised histograms of pixel values constructed for each row of the textrogram are stacked vertically to obtain the anti-spoofing feature vector in the same manner as GMM mean-vectors are stacked to form supervectors. LBP-based features are calculated for the test data and that used for training client models. The two resulting feature vectors are compared using histogram intersection and the resulting score is thresholded to decide if the test signal is genuine speech or a spoofing attack. Experimental results presented in [27] showed that the LBP-based textrogram analysis is effective in detecting a range of spoofed speech signals, including artificial signals [11], [12] and speech synthesis (EERs of below 1%) while less effective in detecting voice conversion (EER in the order of 7%).

IV. EXPERIMENTAL SETUP

The comparison of replay, speech synthesis and voice conversion spoofing attacks requires the sensible choice and adaptation of a common, large and standard corpus and strictly controlled experimental protocols. They are described here.

A. Methodology

to do later...

Due to lack of real replay recordings (e.g., similar to the MOBIO corpus collected in real environment [28]) we had to use artificial setup of replay environment, however, using impulse responses calculated using real playback hardware and real acoustic environments.

This work will also verify the effectiveness of replay detection, using two previously described replay countermeasures:

Case for emulated attacks - there is no alternative. Similar to what has been done for all other work.

In the following we describe the ASV systems used in this study, the datasets, protocols and metrics, and then the implementation of replay emulation and implementation of the countermeasures.

B. ASV systems

Since previous work in ASV anti-spoofing has shown different vulnerabilities for different ASV systems, this work was performed with a range of representative technologies, from the standard to the state-of-the-art. They include: (i) a standard Gaussian mixture model with universal background model (GMM-UBM) system with 1024 Gaussian components; (ii) a GMM supervector linear kernel (GSL) system; (iii) a GSL system with nuissance atribute projection (NAP) [29]; (iv) a GSL system with factor analysis (FA) [30]; (v) a GMM-UBM system with factor analysis; (vi) an iVector system [31] (referred to from here on as IV), and (vii) an iVector system with probabilistic linear discriminant analysis (PLDA) [32] and length normalisation [33] (referred to from here on as IV-PLDA). All seven ASV systems were tested with and without score normalisation. Symmetric normalisation (S-norm) [34] was applied to IV and IV-PLDA systems while test normalisation (T-norm) [35] was used for the others.

All ASV systems were implemented with the same LIA-SpkDet toolkit [36] and the ALIZE library [37] and were directly derived from the work in [30]. They use a common UBM with 1024 Gaussian components and a common feature parametrisation: linear frequency cepstral coefficients (LFCCs), their first derivatives and delta energy. The speech activity detector is also common to each system. It fits a 3-component GMM to the log-energy distribution and adjusts the speech/non-speech threshold according to the GMM parameters [38]. This approach has been used successfully in many independent studies, e.g. [39], [40].

C. Datasets, protocols and metrics

Experiments were performed on the male subsets of the 2004, 2005, 2006 and 2008 NIST Speaker Recognition Evaluation datasets, which are from here on referred to as NIST'0x.
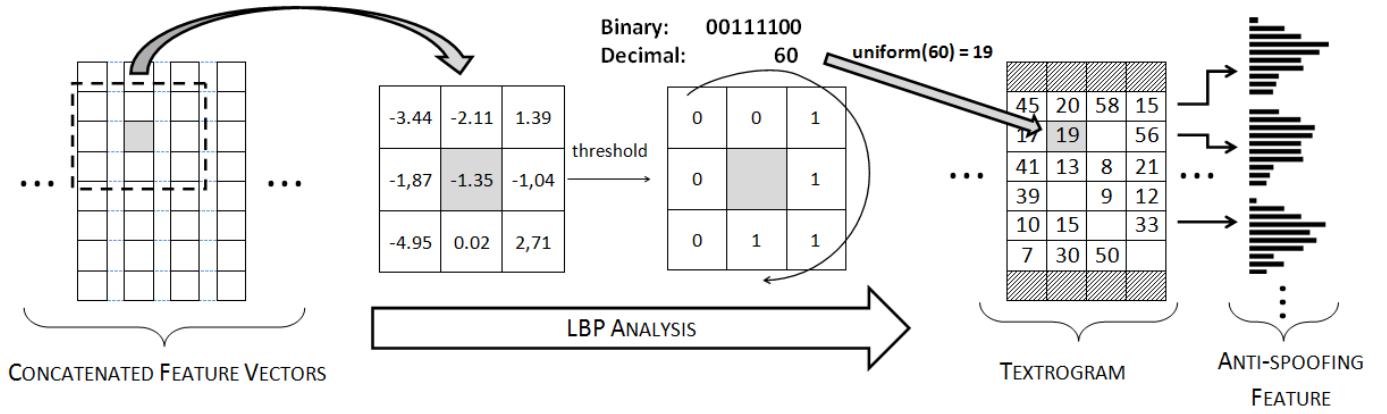
Fig. 2: Schematic diagram of LBP-based feature extraction.

The NIST'04 and NIST'08 datasets were used for UBM training. The NIST'05 dataset was used for development, whereas the NIST'06 dataset was used for evaluation; only results for the latter are reported here. Due to the significant amount of data necessary to estimate the total variability matrix T used with the IV-PLDA system, the NIST'06 dataset was added to the pool of background data for development whereas the NIST'05 dataset was used for evaluation. T is thus learned using approximately 11,000 utterances from 900 speakers, while independence between development and evaluation datasets is always respected.

All experiments relate to the 8conv4w-1conv4w condition where one conversation corresponds to an average of 2.5 minutes of speech (one side of a 5 minute conversation). In all cases, however, only one of the eight, randomly selected training conversations was used for enrolment. Results presented in this paper are thus comparable only to those in the literature for the 1conv4w-1conv4w condition. Standard NIST protocols dictate in the order of 1,000 true client tests and 10,000 impostor tests for development and evaluation datasets. To assess spoofing, impostor tests are replaced with spoofed versions of the original utterance, whereas genuine client trials are unchanged. This setup conforms to the general protocols outlined in [2].

Given the consideration of spoofing, and without any standard operating criteria under such a scenario, the equal error rate (EER) metric is preferred to the minimum detection cost function (minDCF). Also reported is the spoofing false acceptance rate (SFAR, [41]) for a false rejection rate (FRR) fixed to the baseline EER.

D. Spoofing emulation

Spoofing attacks are generally assumed to be performed at the microphone level. In the case of voice conversion and speech synthesis, the practical scenario would then involve the recording of suitable data for the training or adaptation of conversion or synthesis systems, the fabrication of a spoofed utterance and then its presentation to the microphone of the ASV system. None of the past work (including the authors') follows this process, preferring instead to emulate spoofing

attacks by intervening after the microphone, immediately prior to feature extraction. While it may not reflect the practical scenario, the approach can be justified [] and is that adopted here.

Voice conversion spoofing attacks were emulated with the approach described in Section II-B. A worst-case scenario is considered; conversion is performed with full prior knowledge of the ASV system, i.e. voice conversion is performed with exactly the same front-end processing as that used for ASV. The filtering model $H_y(f)$ and filter $H_x(f)$ use 19 LPCC and LPC coefficients respectively. Voice conversion is applied to the original impostor utterances which are converted towards the genuine speaker for any given trial.

Speech synthesis attacks are emulated according to the approach described in Section II-A using the voice cloning toolkit[2] with a default configuration and standard speaker-independent models trained on the EMIME corpus [42]. Adaptation data for each target speaker comprises three utterances (with transcriptions). For any given trial, speech synthesis spoofing attacks are generated using arbitrary text, thereby producing a spoofed utterance of duration close to that of the average test utterance.

Replay attacks are emulated according to the approach described in Section II-C. Emulations include a random mix of three different loudspeakers impulse responses $spk(t)$ and three different replay environments $b(t)$. Speaker impulse responses were obtained from [43] and correspond to a low-quality smartphone speaker, a medium-quality tablet speaker and a high-quality standalone speaker. The impulse response and frequency responses of each are illustrated in Fig. 3. There are significant differences in the frequency responses which show in particular the high-pass functions of the lower quality devices. We decided to consider the worst-case example, i.e., the case when the spoofer possesses a high-quality recording of the target speaker, therefore we decided to ignore the impact of $mic(t)$ and $a(t)$.

The first two replay environment impulse responses were obtained from [44] and correspond to an enclosed medium-sized office and an open corridor. The third impulse response
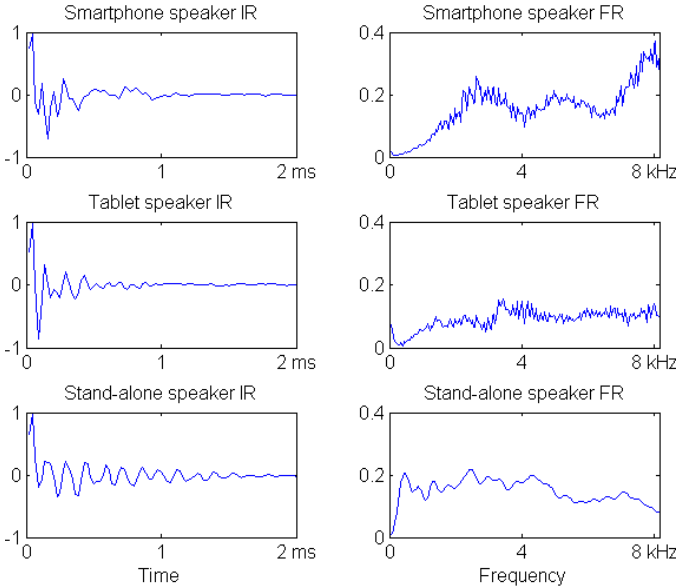
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[2]http://homepages.inf.ed.ac.uk/jyamagis/software/page37/page37.html

Fig. 3: Impulse left) and frequency (right) responses for three different speakers.

simulates an anechoic chamber with a flat frequency response.

One potential problem here: in the use case relevant to this work, voice conversion speech synthesis and replay attacks will all involve replay, and thus a particular $b(t)$, yet your experiments only consider $b(t)$ for replay. The comparison is therefore not fair. I suggest to remove $b(t)$ in your replay work. In any case there is no difference between the two impulse responses.

[AJ] I think this is not always the case. We can easily imagine a VC or SS application running on a smartphone and emitting converted/synthetic voice directly to the telephone channel. Or a hardware gadget which is plugged to a smartphone instead of a headset – there's no replay here either. If replay is involved, efficiency of SS or VC attack is even lower. Without considering $b(t)$ for them we in fact work in favour of them. In the replay attack, in the simplest usage case you don't use any dedicated applications, this is why we have to consider $b(t)$. Besides, the difference between them is remarkable, see the results in Table III and Table II.

E. Countermeasures

The far-field channel detection countermeasure was implemented according to the algorithm originally proposed in [25] and described in Section III-B. The modulation indices were calculated frame-wise from the speech signal envelope which was approximated by the absolute value of the signal after down-sampling to 60 Hz. The final modulation index was calculated by averaging it over the speech frames whose modulation index was above 0.75.

The LBP countermeasure was implemented using the toolkit provided by The University of Oulu[3]. Normalised acoustic features used for LBP analysis were composed of 51 coefficients:

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[3]http://www.cse.oulu.fi/CMV/Downloads/LBPMatlab

16 LFCCs and energy plus their corresponding delta and delta-delta coefficients. Analysis is based on speech frames and using only the 58 so-called uniform LBPs[4] as originally described in [26] and for speech processing in [27]. LBP histograms are created for all but the first and last rows, i.e., for $51 - 2 = 49$ rows. Since we ignore any LBPs but the uniform 58 ones, thereby the output feature vector will be $58 \times 49 = 2842$-elements long.

Both countermeasure algorithms were trained using a random subset of 1000 utterances from the NIST'05 dataset which were treated as described in Section IV-C in order to generate suitable training data with various acoustic conditions. Room and loudspeaker impulse responses (a lecture room, a staircase and a meeting room) different to those used for ASV experiments ensure no countermeasure over-fitting.

A Bayesian network classifier [45] was learned to differentiate genuine data from spoofed data in the case of the far-field channel detection countermeasure. In contrast, the AdaBoost M1 meta classifier [46] was used for the LBP countermeasure. These classifiers returned the best results in each case for an area-under-the-ROC metric.

During evaluation, first, the spoofing detector was used independently from the ASV system, similarly to the protocol used, e.g., in ASVspoof 2015 Challenge evaluation [47]. In our version of the protocol we used 1352 genuine trials and 8112 replay attacks, using various replay hardware and acoustic environments. We ignored the anechoic chamber case, as it is not likely in a real life scenario. In further experiments, we integrated it with an ASV system, as it was done, e.g., in [27]. In this case trials which were classified as spoofs were assigned an arbitrarily low score and were thereby automatically rejected.

V. RESULTS

Attention now turns to the assessment of ASV vulnerabilities to replay spoofing, which are shown to be comparable to those of voice conversion and speech synthesis. These findings motivate the assessment of replay countermeasures.

A. Replay spoofing

Fig. 4 shows the effect of replay attacks on the score distributions for the IV-PLDA system. While the zero-effort impostor and genuine client distributions are well separated, the latter overlaps significantly with the score distribution for replay attacks collected in an office environment. Increased overlap between these distribution will degrade ASV performance.

Fig. 5 illustrates a detection error trade-off (DET) plot[5] for the same experimental setup. The lower-most, solid profile illustrates the performance of the baseline ASV system. The upper-most profile illustrates performance when zero-effort

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[4]The subset of LBPs which contain at most two bitwise transitions from 0 to 1 or 1 to 0 when the bit pattern is traversed in circular fashion

[5]The DETs in this paper were produced with the TABULA RASA Score-toolkit (http://publications.idiap.ch/downolads/reports/2012/Anjos_Idiap-Com-02-2012.pdf) and the Bosaris toolkit (https://sites.google.com/site/bosaristoolkit/).
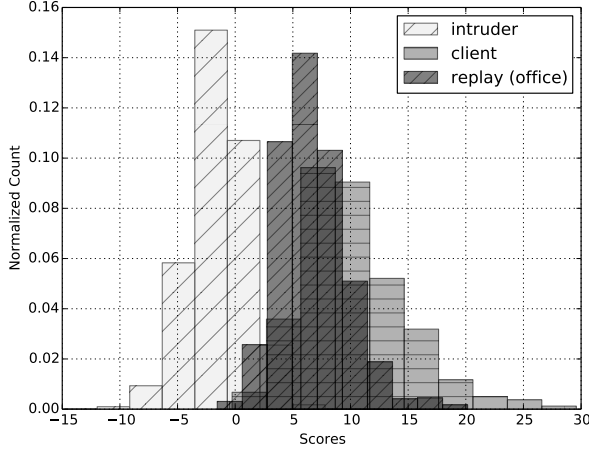
Fig. 4: Score distribution for the IV-PLDA system for replay attacks using a stand-alone speaker and emulation of an office.
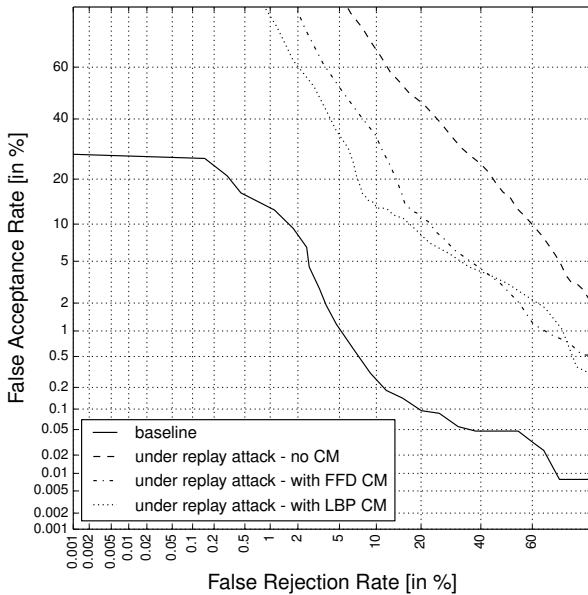

Fig. 5: DET plots for the IV-PLDA system: (i) the baseline, (ii) under replay spoofing with a stand-alone speaker in an office, (iii) under replay attack, but with FFD and (iv) LBP countermeasures - to be referred to later.

impostors are replaced with replay attacks. Thus, the difference between these two profiles serves as an indication of the system vulnerability to replay spoofing; in this case the degradation in performance is significant.

This trend is repeated across the full set of seven ASV systems and different replay attack configurations; results are illustrated in Table II. Results in the second row show the baseline performance for each ASV system and for only zero-effort impostors. As expected, the IV-PLDA system delivers the lowest EER.

Rows 2–5 illustrate the degradation in performance when
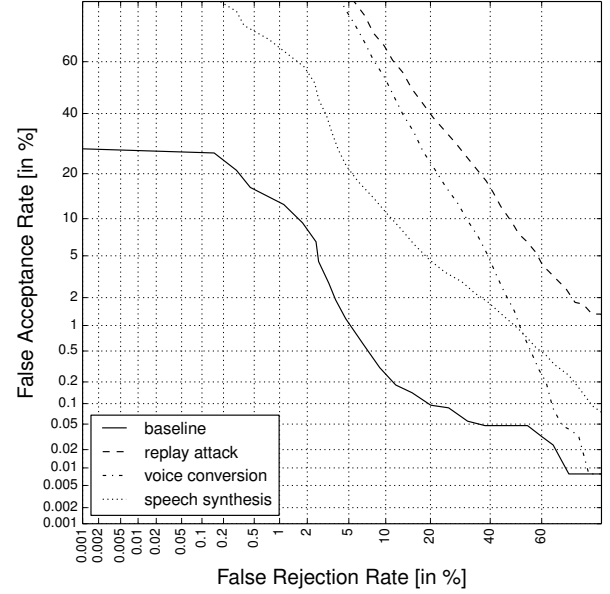

Fig. 6: DET plots for iVector-PLDA system and various attacks.

zero-effort impostors are replaced with replay attacks. EERs are here averaged across the three loudspeaker configurations; other results not reported here showed greater sensitivity to the acoustic environment than to the loudspeaker characteristics. The performance of all seven systems degrades significantly. The EER of the most sensitive GSL system increases from 8% to in the order of 90% for all three acoustic environments. Even the EER of the most resistant SGL-FA system increases to between 22% and 50%. Finally, the EER of the state-of-the-art IV-PLDA system increases to between 25% and 50%.

B. Comparison to voice conversion and speech synthesis

DET profiles showing the vulnerabilities to replay, voice conversion and speech synthesis attacks are illustrated in Fig. 6 for the IV-PLDA system. While the comparison of such profiles is not strictly meaningful[6] it is clear that replay attacks are at least as great a threat to ASV reliability as voice conversion and speech synthesis. This trend is consistent across the full range of ASV systems. Again, while the comparisons are not strictly meaningful, results in rows 6 and 7 of Tab. II show the broad vulnerability of all seven ASV systems to replay attacks. **Can add more here once figures are available.**

C. Replay countermeasure

Fig. 7 illustrates the performance of the far-field and LBP-based countermeasures for replay data collected in an office and a corridor environments. It shows directly the efficiency

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[6]The authors are not experts in speech synthesis, nor voice conversion. Different spoofing algorithms may well yield different comparative results. For example, other authors [48] show considerably greater vulnerabilities for speech synthesis **Do you know other papers with higher EERs for the NIST protocol? Don't we help our reviewers too much here? ;)**

	GMM	SGL	SGL-NAP	SGL-FA	FA	IV	IV-PLDA
Baseline	8.63	8.13	6.31	5.72	5.61	6.72	2.98
Replay: office environment	60.32	92.98	29.92	28.54	30.12	28.89	30.30
Replay: corridor environment	55.91	88.20	23.59	21.62	24.97	23.31	24.53
Replay: anechoic environment	64.40	96.67	49.44	49.31	49.67	49.06	49.46
Voice conversion	33.69	36.92	27.58	23.97	23.96	tbc.	19.30
Speech synthesis	27.29	15.04	13.78	11.91	16.22	tbc.	10.82

TABLE II: Equal error rates (EERs) for seven different ASV systems and for zero-effort impostors (baseline) and three different replay attack configurations (three different acoustic environments). Results are averaged across the three different loudspeaker configurations.

Environment	EER (%)			SFAR (%)		
	no CM	with FFD	with LBP	no CM	with FFD	with LBP
Office	30.30	13.62	9.56	88.70	63.93	46.29
Corridor	24.53	11.34	7.00	80.91	50.25	30.52
None	49.46	42.14	46.77	97.00	95.17	95.76

TABLE III: EER and SFAR values for various environment of replay attacks, with and without the FFD or LBP countermeasures applied, for IV-PLDA. The SFAR was measured for FRR equal to the baseline EER (2.98%).

of spoofing detection, independently from an ASV systems. The plot illustrates that the LBP countermeasure yields much better results than the FFD one, achieving the EER of 2.87%, while the FFD method yielded 16.53%. The step-like shape of the DET curve for the LBP is typical to classifiers based on decision trees.

The next results show performance of the both analysed countermeasures when integrated with the IV-PLDA system. We selected this ASV, since it is the state-of-the-art, widely used ASV solution, and therefore it is likely that it can be a victim of replay spoofing. The middle two profiles in Fig. 5 show the impact of the far-field detection and LBP-based countermeasures on such a system. While they show reductions in the EER compared to the upper-most profile, they remain far from the original baseline.

The detailed results of experiments with FFD and LBP countermeasures for various replay environment, averaged across the replay devices, are presented in Table III. It shows that the countermeasure performance varies depending on acoustic environment. The relative improvement caused by the countermeasures turned out to be the highest for the office – in this case the EER decreased from 30% down to less than 14% for FFD and less than 10% for LBP. Also for the corridor LBP turned out to be more effective than FFD – 7% EER vs. 11%, respectively, also the SFAR result was much lower (30% vs. 46%). When acoustic conditions were not considered, both countermeasures performed poorly, with the EER results slightly better for FFD.

D. Interpretation of results

The results presented above corroborate the findings of previous work performed with considerably smaller databases. While reporting the first work with NIST databases and replay attacks emulated in similar fashion to all past work with voice conversion and speech synthesis, there are some aspects of the
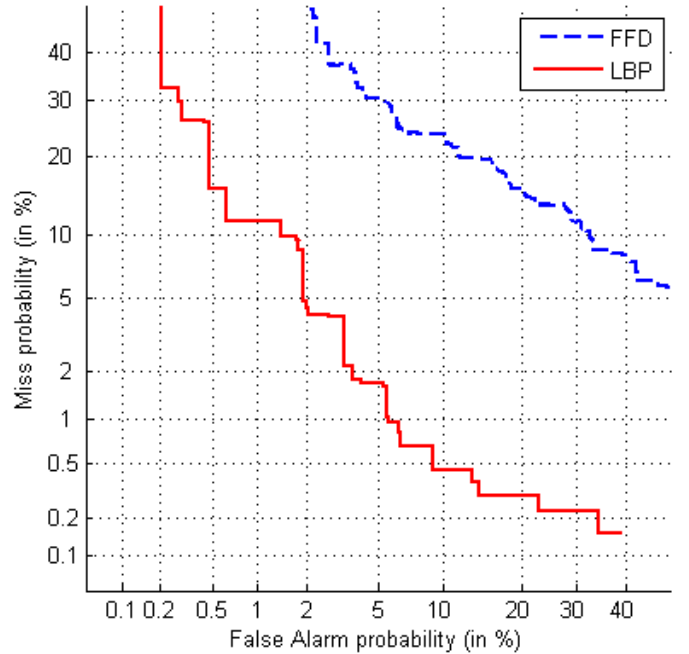

Fig. 7: DET plots for replay spoofing detection using the FFD or LBP countermeasures (measured independently from ASVs).

methodology which must be highlighted in order that results are sensibly interpreted.

First, the results show that replay attacks are of comparable threat to voice conversion and speech synthesis; they are not intended, nor sufficient to show that replay attacks cause the greatest degradations in ASV performance. The relative degradations are naturally a function of the authors' effort and expertise in each approach, whereas they are not expert in voice conversion, nor speech synthesis. Other, perhaps more sophisticated voice conversion and speech synthesis algorithms

may well cause greater degradations in ASV performance than those reported here. The work should therefore be taken as evidence only of the importance of ensuring ASV robustness to replay spoofing in the future.

These arguments lead naturally to the comparisons regarding the effort and expertise involved in the implementation of each spoofing attack. Whereas voice conversion and speech synthesis are relatively high-effort, high-technology attacks, replay spoofing is implemented easily, without any specific expertise or equipment. Accordingly, the threat posed by replay attacks should be considered greater in a practical perspective, even in the case that future work shows degradations in ASV performance cased by replay are lower than those caused by voice conversion and speech synthesis; they are in any case a significant threat and the most likely form of spoofing 'in the wild'.

E. Future work

There are also some aspects of the results which require further work to investigate and explain. Of particular note is the vulnerability of the SGL system whose EER increases to 90% when subjected to replay attacks; all other ASV systems have EERs in the order of 25% to 50%. Our initial investigations have shown that score normalisation may be the cause. Other experiments without score normalisation showed EERs of between 30% and 50%. Accordingly, further work is required to study the impact of score normalisation on ASV vulnerabilities to spoofing.

There are also some as yet unanswered questions regarding the impact of channel compensation. The main idea behind the detection of replay attacks essentially involves the detection of unexpected channels. Thus, while channel compensation has unquestionable utility in ensuring the usability of ASV across different devices, for example, it may also work to the fraudster's advantage; it can make the very channel characteristics needed to prevent replay attacks more difficult to detect. There is some evidence of this in the difference between results for the IV and IV-PLDA systems. While the latter gives better baseline performance, it is also marginally more vulnerable to replay attacks. Further work should investigate this link.

Finally, further work should assess the threat with a more application-driven methodology. That chosen here was motivated by the need to compare the threat of replay to that of voice conversion and speech synthesis, simply as a means of prioritising future work, i.e. to determine whether or not replay is a genuine threat. While the the NIST SRE datasets were essential in order to support comparisons to other work, and are almost a requirement as regards publication, they were designed more for surveillance and security scenarios rather than authentication applications more relevant to spoofing. Since replay spoofing is undeniably application specific, future work should thus consider more the application than the dataset. It is stressed, however, that this criticism can be levelled equally to all of the past work.

VI. CONCLUSIONS

This paper assesses the threat to automatic speaker verification of replay spoofing attacks. The threat is shown to be of least similar significance to that of voice conversion and speech synthesis. The latter have attracted by far the greatest attention in the literature to date. This paper argues that replay spoofing deserves greater attention; in contrast to voice conversion and speech synthesis, they require no specialist expertise or equipment and are therefore the most likely attack from a practical perspective.

While there is potential to detect replay spoofing attacks, the approaches assessed in this paper rely upon the presence of measurable channel effects. Approaches to channel compensation commonly used in today's state-of-the-art speaker verification systems can reduce channel effects thereby increasing the difficulty in detecting replay attacks. Finally, replay attacks recorded with high-quality sound systems will be almost impossible to detect. Alternative countermeasures will then be needed. Challenge-response mechanisms are a suitable candidate but lack scientific and objective validation.

Lastly, the practical use of these and any other replay countermeasure will be dependent on the application. It is extremely difficult to assess replay spoofing, hence the rather bold assumptions in this work and the use of emulated attacks. While this is also a weakness of much of the past work, it is particularly true in the case of replay. Further work will be needed to assess replay, and spoofing in general, with an application-driven methodology. In all of these, there will be potential for replay attacks. Since any security system is only as strong as its weakest link, replay is a genuine threat.

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