UNIVERSITAT POLITECNICA DE CATALUNYA
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Tesi Doctoral

TECNICAS DE SPEECH ENHANCEMENT CONSIDERANDO ESTADISTICAS DE ORDEN SUPERIOR

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CAPITULO VII

Conclusiones.

El objetivo de este último capítulo es el de recopilar las conclusiones obtenidas durante los capítulos precedentes con la intención de extraer unas conclusiones generales sobre el uso de las técnicas de Speech Enhancement presentadas en este trabajo.

Los objetivos esenciales que nos marcamos en esta Tesis Doctoral fueron expuestos en la Introducción: a partir del esquema básico de Filtrado Iterativo de Wiener aplicado al procesado digital de voz, crear un sistema de Speech Enhancement suficientemente robusto y versátil como para resultar efectivo ante las situaciones más adversas y en entornos reales de funcionamiento, tanto como sistema directamente dirigido al oído humano, como preprocesado previo a otras aplicaciones de Tratamiento de Voz.

Se partió del estudio del método AR2 clásico y se han realizado unas primeras incursiones en el campo de las estadísticas de orden superior (AR3 generalizado y AR4 básico). En el presente trabajo, además de optimizar el algoritmo central, se ha realizado el estudio exhaustivo de todos estos métodos, tanto en su formato elemental como de las nuevas variantes introducidas. La inclusión de la figura del Factor Intertrama **IF** en cada una de las técnicas evaluadas es seguramente el paso más importante realizado en términos de complejidad de cálculo, dado lo prometedor de sus resultados. Asimismo, se ha abierto un nuevo camino hacia las técnicas de Reconocimiento de Voz a través de la utilización de los algoritmos iterativos mediante los modelados OSA_AR2 y OSA_AR2_IF de la señal ruidosa, cuando se evalúen entornos altamente ruidosos (SNRG=0dB).

Concretando más sobre el estudio realizado y tomando siempre como punto de referencia al método de correlaciones, el primer camino alternativo que se explotó fue el uso de las HOS, partiendo de sus particulares propiedades según las cuales, en primer lugar, todos los cumulantes de orden superior a dos se anulan para procesos gaussianos, y en segundo lugar, los procesos no gaussianos con p.d.f. simétrica tienen sus cumulantes de orden impar idénticamente nulos. Considerando las características no gaussianas de la voz (sobretodo sus tramas sonoras) y que una distribución gaussiana o simétrica es una buena aproximación del ruido presente en muchos entornos reales, es posible obtener un modelado AR más independiente del ruido si se consideran los cumulantes de la señal en lugar de las correlaciones del método clásico AR2.

Para todos los algoritmos presentados se ha realizado un estudio analítico de convergencia que nos proporciona información acerca de su velocidad de convergencia y del grado de distorsión ocasionado, además de la evolución de comportamiento ante la presencia de distintos niveles de ruido.

Las cotas alcanzadas por los cumulantes de tercer orden, AR3, en comparación al método clásico demuestran unas prestaciones muy superiores, especialmente en el margen donde el nivel de ruido es más elevado (SNR≤10dB), así como un importante incremento de la velocidad de convergencia del algoritmo hacia la estimación del filtro óptimo. Sin embargo, esta mayor agresividad del método, proporcionada por el mayor desacoplo voz-ruido suministrado por las HOS, se paga con una superior distorsión por cada iteración procesada. En consecuencia, se ha concluido que los algoritmos de orden superior deben poner especial énfasis en no procesar más iteraciones de las necesarias. Se trata, por lo tanto, de una potente herramienta especialmente útil en los casos más desfavorables, tanto frente a ruido blanco gaussiano como a ruido real, tal como ha quedado demostrado en las medidas objetivas y en los tests de audición realizados.

Los algoritmos basados en los cumulantes de cuarto orden (AR4), por su parte, gracias a que preservan las componentes simétricas de la señal (son de orden par), nos proporcionan un filtrado no tan agresivo. Así, pese a necesitar alguna iteración más y no obtener medidas de distancia tan buenas como AR3 para SNR bajas y medias, AR4 mantiene un comportamiento mejor y más equilibrado en el resto de situaciones (ambientes poco ruidosos). En cualquier caso, continúa estando por encima de AR2 para cualquier nivel de ruido, por lo que representa una buena alternativa a los cumulantes de tercer orden en aquellos casos en que la señal de voz procesada tenga una fuerte componente simétrica en su función de densidad de probabilidad. No debe olvidarse el importante incremento de coste de cálculo implícito en dichos cumulantes de cuarto orden. Si queremos que el sistema funcione en tiempo real, éste es un factor importante a tener en cuenta.

Una vez comprobado el potencial de las estadísticas de orden superior, el siguiente punto que se abordó fue la consideración de versiones modificadas que permitieran acelerar la velocidad de convergencia y, de este modo, disminuir el coste de cálculo y obtener una menor distorsión. El uso del filtro de Wiener Generalizado basado en los cumulantes nos permite un cierto control sobre el filtro, a través de los parámetros α y β, para controlar su agresividad y su selectividad. Del estudio de las distintas combinaciones α-β posibles resultan una serie de zonas o valles de mínimas medidas por las que podemos optar, dependiendo de si nos interesa un filtrado más suave, pero menos distorsionante, o si interesa un filtrado más agresivo. En definitiva, se trata de una interesante variante del algoritmo que, además de introducir algunas mejoras respecto de los métodos básicos (AR3 y AR4), resulta especialmente resolutivo como herramienta de control del compromiso entre la velocidad de convergencia y la distorsión.

La aplicación del Factor Intertrama (**IF**) ha resultado muy prometedora. Partiendo de la estacionariedad del tracto vocal entre dos tramas consecutivas, se procedió al promediado intertrama de los coeficientes $\mathbf{a_k}$ de dicho modelo, de modo que la estimación correspondiente a la iteración PFI de una determinada trama de señal contribuyera en un factor **1-IF** a la estimación del modelo de su trama inmediatamente posterior. Los resultados obtenidos en todas las pruebas realizadas han confirmado el buen comportamiento de esta técnica, ya sea con AR2 como con AR3, en el margen de SNR's más bajas. Se aprecia un mejor posicionamiento inicial de los formantes de la voz, que deriva en un procesado más rápido y robusto frente al ruido interferente. Tanto es así que además de obtener importantes mejoras en todas las distancias espectrales, logramos, aún en el peor de los casos, reducir el número de iteraciones necesarias para alcanzar la mejor estimación. Los niveles de inteligibilidad de la señal resultante son comparativamente excelentes, y aunque no se logren eliminar completamente todos los espurios musicales que aparecen en el proceso de filtrado, conseguimos en gran medida rebajarlos. Para SNR´s más altas esta técnica no resulta tan necesaria porque la mayor parte de algoritmos permiten atacar estos niveles de ruido inferiores.

Finalmente se ha realizado el estudio del método AR2 en el dominio de la autocorrelación causal, OSA_AR2, cuya sencilla implementación da resultados ciertamente inimaginables con el resto de técnicas, sobretodo en el caso de AGWN. Sin embargo, los tests de audición han demostrado que lo que ganamos en medidas objetivas y calidad de la señal lo perdemos en inteligibilidad, por lo que no será conveniente su uso orientado directamente al oído humano. No obstante, otras aplicaciones del Procesado de Voz quedan abiertas a su posible aplicación, y concretamente los resultados por ahora obtenidos en el campo del Reconocimiento nos permiten ser optimistas al respecto.

A partir de la evaluación de los diversos métodos propuestos, desde los algoritmos básico hasta las variantes implementadas, se ha intentado dar el mejor marco de aplicación para cada uno de ellos y obtener un amplio abanico de posibilidades según la aplicación y entorno considerados. Calificándolos para su uso en un entorno muy específico, se ha creado una base importante para disponer en el futuro de un sistema robusto global adaptativo, apto frente a cualquier tipo de situación y siempre con la mayor eficiencia posible.

Existe un conjunto de posibilidades a presentar como futuras líneas de investigación futura. Ya se han dado los primeros pasos para la implementación de un sistema adaptativo en longitud de trama, de forma que se eviten los cambios bruscos entre fonemas diferentes. Intentando explotar más el camino del promediado intertrama, se han realizado también algunas pruebas promediando directamente los espectros o densidades espectrales de cada trama de señal, tal vez una buena vía hacia un filtrado aún más refinado. Otra posible línea de futuro puede ser la creación de un sistema mixto AR3-AR4 que active uno u otro modelo a lo largo de la señal procesada en función de la componente simétrica de cada trama, que permita corregir el pobre comportamiento de los cumulantes en aquellas tramas donde la skewness toma valores demasiado pequeños. A otro nivel se considera la posibilidad de modelar la señal de voz según un modelado ARMA con ceros y polos.

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