Module Template for New and Revised Modules[1]

Module Code	EE5C04
Module Name	SPEECH & AUDIO ENGINEERING
ECTS Weighting[2]	5 ECTS
Semester taught	Semester 1
Module Coordinator/s	Associate Professor Naomi Harte
Module Learning Outcomes with reference to the Graduate Attributes and how they are developed in discipline	On successful completion of this module, students should be able to: LO1. Describe, in terms of signal processing, the functioning of the human vocal and auditory systems; LO2. Explain the time and frequency characteristics of speech signals and relate this to the acoustic-phonetic structure of speech; LO3. Explain non-uniform frequency bands and metrics such as mel scales and dBA; LO4. Explain the role of binaural hearing in distinguishing the direction of an acoustic source; LO5. Apply machine learning techniques to develop a speech processing application in speech synthesis, speaker identification or other similar domain; LO6. Analyse the function of feature extraction in speech and audio signal processing; LO7. Assess and explore the privacy and ethical issues around developing a new speech technology LO8. Assess speech system performance in a systematic manner and compare to state of the art systems in the literature **Graduate Attributes: levels of attainment** To act responsibly - Attained To think independently - Attained To develop continuously - Attained To communicate effectively - Attained

[1] An Introduction to Module Design from AISHE provides a great deal of information on designing and re-designing modules.
[2] TEP Glossary

Module Content	Speech is the most important and pervasive form of communication. Speech Engineering requires an understanding of the physiology of the human vocal and auditory systems. This understanding informs the signal processing methods that characterise speech signals. These methods include lossless tube models of speech production; time and frequency domain representations of speech; and window characteristics and time/frequency resolution trade-offs. Statistical signal processing methods such as autocorrelation and linear prediction of speech provide the basis of methods for speech synthesis and language processing. The module will introduce the underlying principles in speech technology such as speaker verification and speech synthesis, including how machine learning underpins many speech technology applications **Module Syllabus** Digital signal processing for speech and audio processing • Time and spectral domain filters for speech and audio • Window functions and time-frequency analysis • Linear and nonlinear operators for feature extraction • Pattern classification Speech production, the auditory system and speech perception • Introduction to phonetics and phonology • Models of speech production • Ear physiology, psychoacoustics and speech perception Applications to speech systems • Machine learning techniques for speech processing • Speech synthesis, speaker recognition, biometrics & forensics • Paralinguistics • Data privacy and ethics
Teaching and Learning Methods	The taught component of this module uses a mixture of lectures and formative computer lab sessions. The students are expected to engage in extensive reading of both course texts and relevant literature.

Assessment Details[3] **Please include the following:** • **Assessment Component** • **Assessment description** • **Learning Outcome(s) addressed** • **% of total** • **Assessment due date**	<table><tr><td>Assessment Component</td><td>Assessment Description</td><td>LO Addressed</td><td>% of total</td><td>Week due</td></tr><tr><td>Speech system development</td><td>Individual Assignment</td><td>5,6,8</td><td>40</td><td>Week 10 of Sem 1</td></tr><tr><td>Exam</td><td>Exam</td><td>All</td><td>60</td><td>As per exam timetable</td></tr><tr><td></td><td></td><td></td><td></td><td></td></tr><tr><td></td><td></td><td></td><td></td><td></td></tr></table>
Reassessment Requirements	100% exam based
Contact Hours and Indicative Student Workload[3]	**Contact hours:** **44 timetabled hours** **Independent Study (preparation for course and review of materials):** Weekly study 2 hours = 24 hours **Independent Study (preparation for assessment, incl. completion of assessment):** Individual Assignment = 40 hours, spread over 6 weeks
Recommended Reading List	**Main text:** Theory and Applications of Digital Speech Processing Lawrence R. Rabiner, Ronald W. Schafer **Extra reading in library:** Fundamentals of Speech Recognition Lawrence Rabiner, B H Juang Speech Synthesis and Recognition, 2nd Edition John N. Holmes, Wendy J. Holmes Principles of Digital Audio, 6th Edition, Ken Pohlmann, McGraw-Hill Applied Speech and Audio processing, Ian McLaughlin, Cambridge University Press, 2009 Speech and Audio Signal Processing: Processing and Perception of Speech and Music, 2nd Edition, Ben Gold, Nelson Morgan, Dan Ellis, Wiley Other papers will be given as assigned reading during the course.
Module Pre-requisite	EE4C5 (or equivalent Digital Signal Processing module)

[3] TEP Guidelines on Workload and Assessment

Module Co-requisite	
Module Website	On Blackboard
Are other Schools/Departments involved in the delivery of this module? If yes, please provide details.	No
Module Approval Date	
Approved by	
Academic Start Year	
Academic Year of Date	