

## **KOLLOQUIUM**

Institut für Elektrotechnik, Elektronik und Informationstechnik

## **Audio Source Separation based on Independent Component Analysis**

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Diskussionsleitung: Prof. Dr.-Ing. W. Kellermann

This talk introduces the blind source separation (BSS) of convolutive mixtures of acoustic signals, especially speech. A statistical and computational technique, called independent component analysis (ICA), is examined. By achieving nonlinear decorrelation, nonstationary decorrelation, or time-delayed decorrelation, we can find source signals only from observed mixed signals. Particular attention is paid to the physical interpretation of BSS from the acoustical signal processing point of view. Frequency-domain BSS is shown to be equivalent to two sets of frequency domain adaptive microphone arrays, i.e., adaptive beamformers (ABFs). Although BSS can reduce reverberant sounds to some extent in the same way as ABF, it mainly removes the sounds from the jammer direction. This is why BSS has difficulties with long reverberation in the real world. If sources are not "independent," the dependence results in bias noise when obtaining the correct separation filter coefficients. Therefore, the performance of BSS is limited by that of ABF. Although BSS is upper-bounded by ABF, BSS has a decisive advantage over ABF: BSS can be regarded as an intelligent version of ABF in the sense that it can adapt without any information on the array manifold or the target location, and several sources can be simultaneously active in BSS.