# BEAMFORMERS COMPARISON CRITERIA

**Jan Ingerle**

Department of Circuit Theory, Faculty of Electrical Engineering,
Czech Technical University, Technická 2, 166 27 Praha 6,
phone: +420 2 2435 2048, fax: +420 2 2431 0784
e-mail: xingerle@fel.cvut.cz

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**Abstract**
It is necessary to have well defined comparison criteria to analyse systems. This article summarises frequently used criteria to compare multi-channel beamformers and describe the features which can be compared by these criteria. Some basic terms are presented to explain particular criteria. Noise reduction and speech enhancement application are supposed.

**Criteria**

The first criterion is the type of noise which the beamformer is able to suppress. Obviously noise is distinguished by its coherence function defined as

$$\Gamma(f) = \frac{\Phi_{ij}(f)}{\sqrt{\Phi_{ii}(f)\Phi_{jj}(f)}}, \tag{1}$$

where $\Phi_{ii}$ is the power spectrum density of $i$-th channel signal and $\Phi_{ij}$ is the cross power density spectrum of $i$-th and $j$-th channel signals. Main noise types are: incoherent, coherent and diffuse. Ideally, the coherence function of incoherent noise equals zero, of coherent noise equals one and of diffuse noise equals sample function. The first two noise types are basic and they are used to analyse fundamental behaviour of the systems. Diffuse noise rises in reverberation rooms and can be used to model reverberation environment behaviour.

The second very useful characteristic is a directivity pattern. The directivity pattern is the signal reduction coefficient as a function of the arriving angle. It shows ability to reduce coherent noise arriving from non-look directions. The specification of this characteristic is a ratio of the signal power from the look direction (a direction of steered beam) to the signal power from the appointed non-look direction.

Next characteristic — a directivity index is bounded up with the directivity pattern. The directivity index, marked in [2] as B, is defined as a ratio of the signal power from the look direction to the average power received from other directions. This parameter is equivalent to the directivity pattern but is used for a diffuse noise field.

Next important characteristic is Log Area Ratio (LAR) defined as

$$LAR(m) = \frac{1}{L}\sum_{l=1}^{L}\left|20\log\frac{g_r(l;m)}{g_t(l;m)}\right|, \tag{2}$$

where $g_r(l;m)$ and $g_t(r;m)$ represent the $l$-th area ratio function of the reference data and the test data respectively computed over the frame ending at time $m$. The area ratio function is defined as:

$$g(l;m) = \frac{1 + k(l;m)}{k(l;m)}, \tag{3}$$

where $k(l;m)$ are $l$ PARCOR coefficients computed over the fame ending at time $m$ which are calculated from the LPC analysis of order $L = 12$. The mean LAR distance is computed between the beginning and the ending point of the test word. The LAR is $l_1$ norm (distance) correlated with the subjective sensation of a testing person so it can partially replace the subject tests. Lower LAR means higher speech quality.

Another criterion is a steering error behaviour. This criterion is interesting because it touches upon typical practical problem: it characterizes behaviour of the system when the look direction is not steered to the speech source properly. It is necessary to do a sophisticated analysis of the particular system to determine the steering error.

It is useful to know also other characteristics to compare multi-channel systems: e.g. a geometry of the array, a necessary number of channels or a computing load. These characteristics are useful especially when an implementation is actual.

Many of the mentioned system parameters are multidimensional — they are not just an one variable function but most of them are a function of time, frequency, signal to noise ratio, angle of coming signals and type of noise. It is necessary to decide which functions is useful to analyse.

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