Lecture IX

Independent component analysis for linear dynamic (convolutive) problems

Ivica Kopriva

Ruđer Bošković Institute

e-mail: ikopriva@irb.hr ikopriva@gmail.com Web: http://www.lair.irb.hr/ikopriva/

Course outline

- Motivation with illustration of applications (lecture I)
- Mathematical preliminaries with principal component analysis (PCA)? (lecture II)
- Independent component analysis (ICA) for linear static problems: information-theoretic approaches (lecture III)
- ICA for linear static problems: algebraic approaches (lecture IV)
- ICA for linear static problems with noise (lecture V)
- Dependent component analysis (DCA) (lecture Vf)

Course outline

- Underdetermined blind source separation (BSS) and sparse component analysis (SCA) (lecture VII/VIII)
- Nonnegative matrix factorization (NMF) for determined and underdetermined BSS problems (lecture VIII/IX)
- BSS from linear convolutive (dynamic) mixtures (lecture X/XI)
- Nonlinear BSS (lecture XI/XII)
- Tensor factorization (TF): BSS of multidimensional sources and feature extraction (lecture XIII/XIV)

Seminar problems

- 1. Blind separation of two uniformly distributed signals with maximum likelihood (ML) and AMUSE/SOBI independent component analysis (ICA) algorithm. Blind separation of two speech signals with ML and AMUSE/SOBI ICA algorithm. Theory, MATLAB demonstration and comments of the results.
- 2. Blind decomposition/segmentation of multispectral (RGB) image using ICA, dependent component analysis (DCA) and nonnegative matrix factorization (NMF) algorithms. Theory, MATLAB demonstration and comments of the results.
- 3. <u>Blind separation of acoustic (speech) signals from convolutive dynamic</u> <u>mixture</u>. **Theory, MATLAB demonstration and comments of the results**.

Seminar problems

- 4. Blind separation of images of human faces using ICA and DCA algorithms (innovation transform and ICA, wavelet packets and ICA) **Theory, MATLAB** demonstration and comments of the results.
- 5. Blind decomposition of multispectral (RGB) image using sparse component analysis (SCA): clustering + L_p norm (0) minimization. Theory, MATLAB demonstration and comments of the results.
- 6. Blind separation of four sinusoidal signals from two static mixtures (a computer generated example) using sparse component analysis (SCA): clustering + L_p norm (0) minimization in frequency (Fourier) domain.**Theory, MATLAB demonstration and comments of the results.**

Seminar problems

- 7. Blind separation of three acoustic signals from two static mixtures (a computer generated example) using sparse component analysis (SCA): clustering + L_p norm (0) minimization in time-frequency (short-time Fourier) domain. Theory, MATLAB demonstration and comments of the results.
- 8. Blind extraction of five pure components from mass spectra of two static mixtures of chemical compounds using sparse component analysis (SCA): clustering a set of single component points + L_p norm (0) minimization in m/z domain. Theory, MATLAB demonstration and comments of the results.
- 9. Feature extraction from protein (mass) spectra by tensor factorization of disease and control samples in joint bases. Prediction of prostate/ovarian cancer. Theory, MATLAB demonstration and comments of the results.

Blind source separation

A theory for blind signal recovery from multichannel observation requiring minimum of *a priori* information.

Problem: X=AS $X \in \mathbb{R}^{N \times T}$, $A \in \mathbb{R}^{N \times M}$, $S \in \mathbb{R}^{M \times T}$

Goal: find A and S based on X only.

Solution $X=AT^{-1}TS$ must be characterized with $T=P\Lambda$ where P is permutation and Λ is diagonal matrix i.e.: $Y \cong P\Lambda S$

A. Cichocki, S. Amari, "Adaptive Blind Signal and Image Processing," John Wiley, 2002.

Independent component analysis (ICA)

• Number of mixtures *N* must be greater than or equal to *M*.

•source signals $s_i(t)$ must be statistically independent.

$$\mathbf{p}(\mathbf{s}) = \prod_{m=1}^{M} p_m(s_m)$$

•source signals $s_m(t)$, except one, must be non-Gaussian.

$$\left\{C_n(s_m) \neq 0\right\}_{m=1}^M \quad \forall n > 2$$

•mixing matrix **A** must be nonsingular.

$$\mathbf{W} \cong \mathbf{A}^{-1}$$

A. Hyvarinen, J. Karhunen, E. Oja, "Independent Component Analysis," John Wiley, 2001. 8/44

In many situations related to acoustics and data communications we are confronted with multiple signals received from a multipath mixture. Sometimes, this is known under the popular name of *cocktail- party* problem.

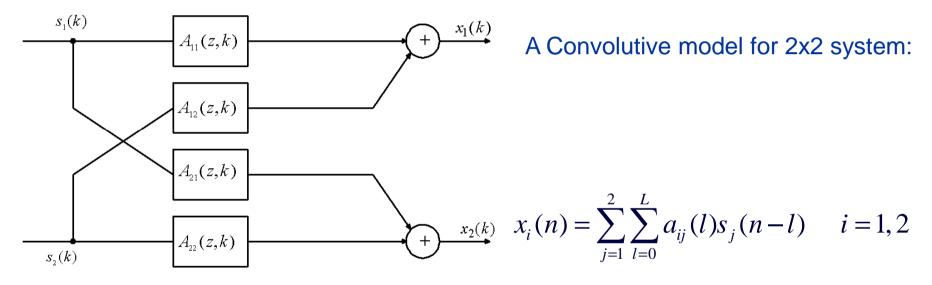
The instantaneous linear mixture model x=As is not valid anymore. Instead, the convolutive model needs to be assumed:

$$\mathbf{x}_{i}(n) = \sum_{j=1}^{M} \sum_{l=0}^{L} a_{ij}(l) s_{j}(n-l) \quad i = 1...N$$

Thus, convolutive mixture is described by a mixing matrix whose elements are the individual impulse responses (in time domain) or transfer functions (in frequency domain) between a source and a sensor.

ICA for convolutive mixtures

When both mixing matrix and sources are unknown the problem is referred to as the multichannel blind deconvolution (MBD) problem¹⁻⁵.



1. A. Hyvarinen, J. Karhunen and E. Oja, *Chapter 19* in Independent Component Analysis, J. Wiley, 2001.

2. A. Cichocki, S. Amari, *Chapter 9* in Adaptive Blind Signal and Image Processing – Learning Algorithms and Applications, J. Wiley, 2002.

3. M. Castella, A. Chevreuil, J.-C. Pesquet, *Chapter 8* in Handbook of Blind Source Separation, Academic Press, P. Comon and Ch. Jutten editors, 2010.

4. R. H. Lambert and C.L. Nikias, *Chapter 9* in Unsupervised Adaptive Filtering – Volume I Blind Source Separation, S.Haykin, ed., J. Wiley, 2000.

5. S.C. Douglas and S. Haykin, *Chapter 3* in Unsupervised Adaptive Filtering – Volume II Blind Deconvolution, S. Haykin, ed., J. Wiley, 2000.

ICA for convolutive mixtures

Multichannel Blind Deconvolution

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Blind Source Separation + Single Channel Blind Deconvolution

solves interchannel interference

solves intersymbol interference

Single channel blind deconvolution problem is also referred to as blind equalization. The unknown source signal that is input to a single-input-singleoutput (SISO) system is required to be i.i.d. non-Gaussian signal. In that case it can be recovered up to indeterminacies: scaling by a constant and delay by a constant i.e.

$$\hat{s}(n) = cs(n - \Delta) \quad c \in \mathbb{C}, \ \Delta \in \mathbb{N}$$

Statistically independent but temporally dependent source signals (audio signals for example) can be recovered blindly only up to permutation and <u>scalar filtering</u> indeterminacies. That is in strong contrast to instantaneous mixtures that for the same scenario allow recovery of the sources up to the scaling by a constant indeterminacy.

The scalar filtering indeterminacy is a consequence of a structure of temporally dependent time series that can be modeled as a convolution of a filter h_i with an i.i.d. driving sequence ε :

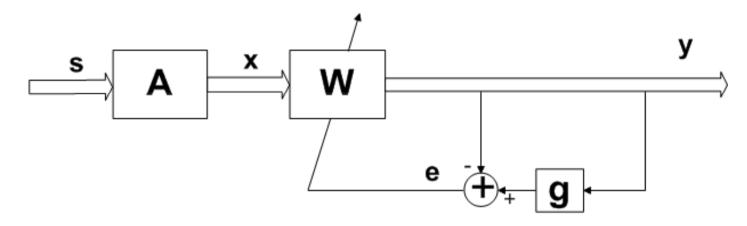
$$s_i(n) = \sum_{p=0}^{i} h_i(p) \mathcal{E}(n-p)$$

Thus, complete deconvolution would destroy a "color" of the signals. Thus, spatial separation between the sources is enough.

In multichannel data communication systems both separation and deconvolution are required. However, all the sources must be i.i.d. non 2/44 Gaussian signals.

ICA for convolutive mixtures

Blind multichannel inverse modeling, equalization and separation: the input signal is unknown and there is no reference or desired signal.



In relation to instantaneous BSS problem, elements of the mixing matrix **A** in convolutive model are filters a_{ij} . They contain impulse responses between the j^{th} input and i^{th} output. We shall assume the number of inputs and outputs to be the same and equal to N.

ICA for convolutive mixtures

Notation and Z-transform preliminaries. NxN mixing matrix is described as:

$$\mathbf{A}(z) = \sum_{n=-\infty}^{\infty} \mathbf{A}_n z^{-n} = \begin{bmatrix} a_{ij}(z) \end{bmatrix}$$
$$a_{ij}(z) = \sum_{n=-\infty}^{\infty} a_{ij,n} z^{-n} \quad i, j = 1...N.$$

It is assumed that each channel is stable i.e. $\sum_{n=-\infty}^{\infty} |a_{ij,n}| < \infty$.

A(z) is a *polynomial matrix* or *Laurent-series matrix* (a matrix whose elements are polynomials, power series or Laurent series).

 \mathbf{A}_{n} is coefficient of the matrix polynomial or matrix Laurent series (a polynomial or Laurent series whose coefficients are matrices).

Convolutive model in *z*-transform domain yields:

$$\mathbf{x}(z) = \mathbf{A}(z)\mathbf{s}(z)$$

where source, measured and recovered signals are described by two-sided *z*-transform: $\sum_{n=1}^{\infty} \sum_{n=1}^{\infty} \sum$

$$s_i(z) = \sum_{n = -\infty}^{\infty} s_{i,n} z^n$$
$$x_i(z) = \sum_{n = -\infty}^{\infty} x_{i,n} z^{-n}$$
$$y_i(z) = \sum_{n = -\infty}^{\infty} y_{i,n} z^{-n} \quad i = 1...N$$

The inverse system is described with:

$$\mathbf{W}(z) = \sum_{n=-\infty}^{\infty} \mathbf{W}_n z^{-n} = \left[w_{ij}(z) \right]$$
$$w_{ij}(z) = \sum_{n=-\infty}^{\infty} w_{ij,n} z^{-n} \quad i, j = 1...N$$
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ICA for convolutive mixtures

Real channels are causal and have finite order L

$$\mathbf{A}(z) = \sum_{n=0}^{L} \mathbf{A}_{n} z^{-n} = \left[a_{ij}(z) \right]$$

Channel order *L* has to be estimated and is related to the maximal delay τ_{max} that can occur in the multipath scenario:

$$L \ge \tau_{\max} F_s$$

where F_s represents sampling frequency. However, noncausal representation is necessary to model inverse of the non-minimum phase (NMP) channels:

$$\mathbf{W}(z) = \sum_{n=-L}^{L} \mathbf{W}_{n} z^{-n} = \begin{bmatrix} w_{ij}(z) \end{bmatrix}$$
$$w_{ij}(z) = \sum_{n=-L}^{L} w_{ij,n} z^{-n} \quad i, j = 1...N.$$
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ICA for convolutive mixtures

Reconstructed signals are obtained as:

$$y_i(t) = \sum_{j=1}^{N} \sum_{l=-L}^{L} w_{ij,l} x_j(t-l)$$
 $i = 1...N$

Noncausal implementation of the inverse systems requires that measured signals **are known ahead in time**. Because this can not be realized a delay line of *L* samples is introduced in order to realize noncausal implementation:

$$y_i(t-L) = \sum_{j=1}^{N} \sum_{l=-L}^{L} w_{ij,l} x_j(t-L-l) \quad i = 1...N$$

This implies delay of *L* samples independently on whether adaptive or blockadaptive implementations are used.

ICA for convolutive mixtures

Why stable inverse of NMP system requires non-causal implementation?

Consider a first order transfer function:

$$w_{ij}(z) = \frac{1}{1 - az^{-1}}$$

for some real *a*. $w_{ij}(z)$ is consider to be inverse of the direct filter $a_{ij}(z)=1-az^{-1}$. Region of convergence (ROC) is given with:

Let us assume that |a|<1 in which case poles of $w_{ij}(z)$ lies inside the unit circle for $z=exp(j\omega)$ and $w_{ij}(z)$ can be represented by causal (one-sided) *z*-transform:

$$w_{ij}(z) = \frac{1}{1 - az^{-1}} = \sum_{n=0}^{\infty} a^n z^{-n}$$
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Let us now assume that |a|>1 in which case poles of $w_{ij}(z)$ lies outside the unit circle for $z=exp(j\omega)$. In this case the ROC becomes

|z| < |a|

A sequence with z-transform $w_{ij}(z)=1/(1-az^{-1})$ and above ROC is given with:

$$w_{ij}(n) = -a^n u(-n-1)$$

where u() represents step function. *z*-transform of $w_{ii}(n)$ can now be written as:

$$w_{ij}(z) = \sum_{n=-\infty}^{\infty} w_{ij}(n) z^{-n} = \sum_{n=-\infty}^{\infty} -a^n u(-n-1) z^{-n} = -\sum_{n=-\infty}^{-1} a^n z^{-n}$$

which represents stable (1/|a|)<1 but non-causal *z*-transform necessary to implement inverse of the NMP channel $a_{ii}(z)=1-az^{-1}$.

ICA for convolutive mixtures

Single channel blind deconvolution.

$$y(t) = \sum_{l=-L}^{L} w_l x(t-l)$$
$$x(t) = \sum_{l=0}^{L} a_l s(t-l)$$

Assumptions:

A1) Source signal must be non-Gaussian. Due to the central limit theorem x(t) is very close to Gaussian process even if s(t) is non-Gaussian. Maximizing departure from Gaussianity does not make sense if source signal is Gaussian.

A2) Source signal must be independent identically distributed (i.i.d.) process (temporally white):

$$E[s(t)s(t-l)] = \sigma \delta_l$$

$$\mathbf{p}(s(t), s(t-l)) = p(s(t)) p(s(t-l)) \lor l > 0$$

$$p_s(t_1) = p_s(t_2)$$

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ICA for convolutive mixtures

Single channel blind deconvolution problem as instantaneous ICA problem:

$$\tilde{\mathbf{x}} \cong \mathbf{A}\tilde{\mathbf{s}}$$

$$\tilde{\mathbf{s}}(t) = \begin{bmatrix} s(t) \ s(t-1) \dots \ s(t-2L+1) \end{bmatrix}^T \qquad \mathbf{A} = \begin{bmatrix} a_1 \ a_2 \dots a_L \\ 0 \ a_1 \ a_2 \dots a_{L-1} \\ \dots \\ 0 \dots \\ 0 \dots \\ 0 \dots \\ 0 \dots \end{bmatrix}$$

ICA independence assumption implies:

E[s(t)s(t-1)...s(t-L)] = E[s(t)]E[s(t-1)]...E[s(t-L)] $\mathbf{p}(s(t), s(t-1), ..., s(t-L)) = p(s(t))p(s(t-1))...p(s(t-L))$

that is equivalent to i.i.d. assumption.

ICA for convolutive mixtures

Why colored signals can not be completely deconvolved?

Z-transform of the white (i.i.d.) source signal:

$$S(z) = \sigma_s^2 z^{-\Delta_s}$$

Z-transform of the colored source signal:

$$S(z) = \sigma_s^2 D(z) z^{-\Delta_s}$$

$$D(z) = \sum_{p} d_{p} z^{-p}$$

Z-transform of the reconstructed signal:

$$Y(z) = W(z)A(z)S(z)$$

Minimization of statistical independence between the sources (spatial separation) yields sources:

$$\hat{S}_{i}(z) = P_{i}(z)S_{i}(z) = \sigma_{s_{i}}^{2}P_{i}(z)D_{i}(z)z^{-\Delta_{s_{i}}} \quad i = 1,...,N$$

That is because if : $\{s_i(n)\}_{i=1}^N$ are independent so are $\{\hat{s}_i(n)\}_{i=1}^N$.

Thus, exploiting statistical independence assumption only, the non-i.i.d. (color) sources can be recovered up to the scalar filtering indeterminacy only.

To, possibly, remove $P_i(z)$ further processing of $\{\hat{s}_i(n)\}_{i=1}^N$ is necessary. Since, $\{\hat{s}_i(n)\}_{i=1}^N$ are statistically independent the only redundancy that is left is temporal dependence within each $\{\hat{s}_i(n)\}_{i=1}^N$

Goal in single channel blind deconvolution:

$$Y(z) \cong S(z) z^{-\Delta}$$

Unsupervised learning criteria are based on the maximization of independence between samples of the sequence y(t):

$$f_{y}(y(t), y(t-1), ..., y(t-m+1)) = \prod_{i=0}^{m-1} f(y(t-i))$$

this will yield:

$$E[y(t-i)y(t-j)] = \sigma_y^2 \delta_{ij}$$

which implies:

$$Y(z) = \sigma_y^2 z^{-\Delta_y}$$

which further implies that: $P_i(z)D_i(z) \rightarrow \sigma_y^2 z^{-\Delta_{y_i}}$ i=1,...,N

i.e. colored signals can not be recovered in completely blind scenario.^{24/44}

Multichannel blind deconvolution. The same assumptions extend to multichannel blind deconvolution (MBD=BSS+SBD):

A1) all source signals must be non-Gaussian.A2) all source signals must be statistically independent and i.i.d. processes:

$$E\left[s_{i}(t-r)s_{j}(t-q)\right] = \sigma^{2}\delta_{ij}\delta_{rq}$$

$$p\left(s_{i}(t-r)s_{j}(t-q)\right) = p\left(s_{i}(t-r)\right)p\left(s_{j}(t-q)\right) \forall r \neq q$$

General solution of the blind source separation problem is given with:

$$\mathbf{G}(z) = \mathbf{W}(z)\mathbf{A}(z) = \mathbf{P}\mathbf{\Lambda}\mathbf{D}(z)$$

Where **P** is general permutation matrix, Λ is diagonal scaling matrix and

$$\mathbf{D}(z) = diag\left\{d_1 z^{-\Delta_1} \dots d_N z^{-\Delta_N}\right\}$$

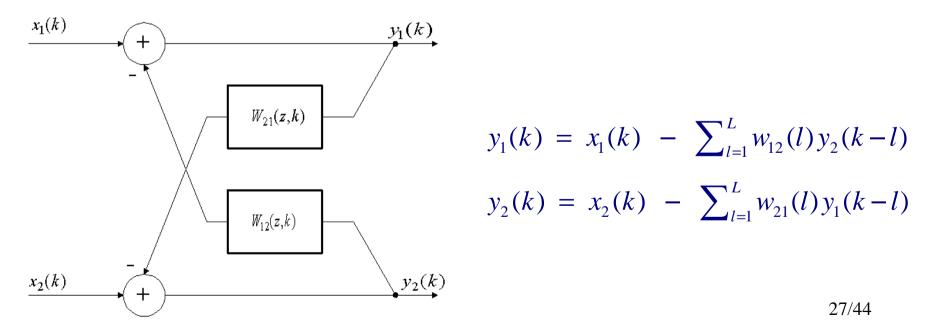
Objective of the BSS in the most general case is to recover possibly scaled, reordered and filtered estimates of the unknown source signals.

- Multichannel blind separation and deconvolution in time domain
 - (+) On-line formulation is realizable.
 - (-) Slow convergence.
 - (-) Colored signals are whitened if feedforward architecture is used.
 - (-) Whitening can be avoided with feedback architecture but this prevents implementation of the non-causal inverse filters required to implement stable inverse of the NMP mixing channels.
- Multichannel blind separation and deconvolution in frequency domain
 - (+) Convergence is faster due to the fact that frequency bins are orthogonal.
 - (+) Condition for non-causal implementation is satisfied naturally through block filtering implementation.
 - (-) Only block-adaptive but not truly adaptive implementation can be achieved.
 - (-) Permutation on the frequency bin levels causes serious difficulties when signals have to be transformed back in time domain.
 - (-) Computationally more complex to implement.

Time domain approach with recurrent (feedback) architecture.

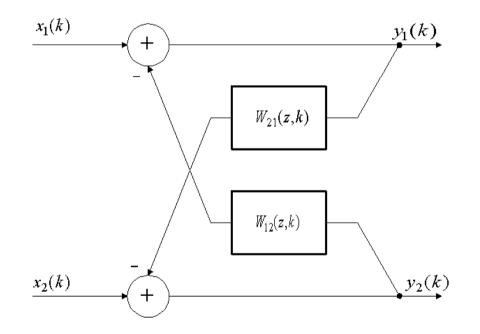
(+) Whitening effect is eliminated due to the fact that signals $y_i(t)$ have to have temporal structure in order to cancel appropriate source signal $s_j(t)$ in the mixture $x_n(t)$.

(-) It is not possible to realize non-causal implementation necessary to invert non-minimum phase mixture channels.

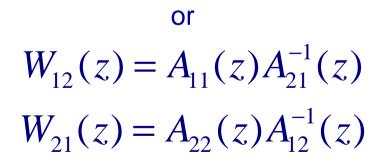


ICA for convolutive mixtures

Time domain approach with recurrent (feedback) architecture. Asymptotic solutions for unmixing filters.



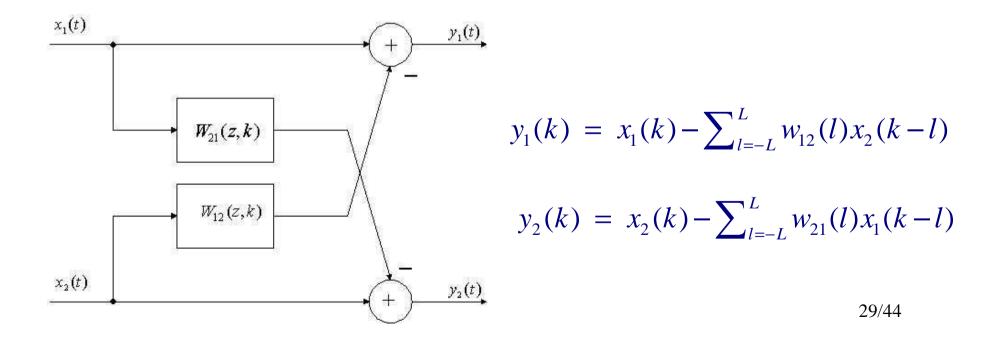
 $W_{12}(z) = A_{12}(z)A_{22}^{-1}(z)$ $W_{21}(z) = A_{21}(z)A_{11}^{-1}(z)$



Time domain approach with feedforward architecture.

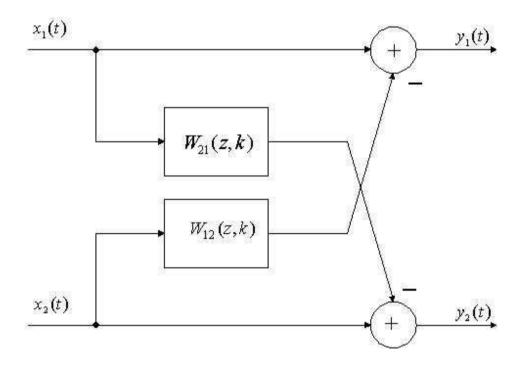
(+) Non-causal realization necessary to invert non-minimum phase mixtures is easily implemented by pure delay line.

(-) Direct filters set to unity only partially eliminate the whitening effect.



ICA for convolutive mixtures

Time domain approach with feedforward architecture. Asymptotic solutions for unmixing filters.



 $W_{12}(z) = -A_{12}(z)A_{22}^{-1}(z)$ $W_{21}(z) = -A_{21}(z)A_{11}^{-1}(z)$ or $W_{12}(z) = -A_{11}(z)A_{21}^{-1}(z)$ $W_{21}(z) = -A_{22}(z)A_{12}^{-1}(z)$

Contrast function for MBD problems are in a majority of cases derived or can be related to maximum-likelihood formulation of the MBD problem^{5,6}.

$$J_{ML}(\mathbf{W}) = \int_{\mathbf{x}} f_{\mathbf{x}}(\mathbf{x}) \log \hat{f}_{\mathbf{x}}(\mathbf{x}, \mathbf{W}) d\mathbf{x} = E \left\{ \log \hat{f}_{\mathbf{x}}(\mathbf{x}, \mathbf{W}) \right\}$$

that amounts to minimizing spatial and temporal independence. Noting that $\hat{f}_{\mathbf{x}}(\mathbf{x}, \mathbf{W}) = \left[\hat{f}_{\mathbf{x}}(\mathbf{x}, \mathbf{W}) / f_{\mathbf{x}}(\mathbf{x}) \right] f_{\mathbf{x}}(\mathbf{x})$

likelihood contrast function can be written as

$$J_{ML}(\mathbf{W}) = -D\left(f_{\mathbf{x}} \| \hat{f}_{\mathbf{x}}\right) - H(f_{\mathbf{x}})$$

where D() represents Kullback-Leibler distance.

⁶D.L. Donoho, "On minimum entropy deconvolution," in D.F. Findley, ed., *Applied Time Series* 31/44 *Analysis II,* Academic Press, pp.565-608, 1981.

ICA for convolutive mixtures

Learning rule for W is obtained as:

$$\Delta \mathbf{W} \propto \frac{\partial J_{ML}(\mathbf{W})}{\partial \mathbf{W}} = -\frac{\partial D(f_{\mathbf{x}} \| \hat{f}_{\mathbf{x}})}{\partial \mathbf{W}}$$

which means that maximization of likelihood is equivalent to minimization of distance between true unknown and model pdf's for a set of measurement.

When MBD learning rules are derived by maximizing likelihood function they will contain score functions as nonlinear functions:

$$\varphi_i(y_i) = -\frac{1}{f_{s_i}(y_i)} \frac{df_{s_i}(y_i)}{dy_i}$$

ICA for convolutive mixtures

As shown before parameterized form of the score functions can be derived from generalized Gaussian distribution:

$$\varphi_i(y_i) = sign(y_i) |y_i|^{\alpha_i - 1}$$

With the single parameter α_i (called Gaussian exponent) super-Gaussian distributions ($\alpha_i < 2$) and sub-Gaussian distributions ($\alpha_i > 2$) could be modeled.

If MBD is applied in communication environment $\alpha_i=3$ is good choice yielding:

$$\varphi_i(y_i) = sign(y_i) |y_i|^2$$

If MBD is applied on audio signals $\alpha_i = 1$ is good choice yielding:

$$\varphi_i(y_i) = sign(y_i)$$
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On-line (time domain) *Infomax* or maximum likelihood ICA algorithm for blind separation of convolved minimum phase mixtures with feedback architecture. Colored signals such as speech could be separated.

$$\Delta w_{ij}(t,l) \cong \frac{\partial H(\varphi(\mathbf{y}))}{\partial w_{ij}(l)} = \varphi_i(y_i) y_j(t-l)$$
$$w_{ij}(t+1,l) = w_{ij}(t,l) + \mu \Delta w_{ij}(t,l)$$
$$y_i(t) = x_i(t) - \sum_{\substack{j \neq i \\ j=1}}^N \sum_{l=1}^L w_{ij}(t,l) y_j(t-l)$$

where *i*,*j* denote signal indices, *t* denotes iteration index, *l* denotes coefficient index and μ represents learning gain.

Frequency domain approach to MBD.

$$\mathbf{X} = FFT[\mathbf{x}]$$

$$\mathbf{W}_{k}(l+1) = \mathbf{W}_{k}(l) + \left[\mathbf{I} - \mathbf{\Phi}(\mathbf{Y}_{k})\mathbf{Y}_{k}^{H}\right]\mathbf{W}_{k}(l)$$

$$\mathbf{Y}_{k} = \mathbf{W}_{k}\mathbf{X}_{k}$$

$$\mathbf{y} = IFFT[\mathbf{Y}]$$

Where *k* denotes frequency bin index and *l* denotes iteration index. Permutation indeterminacy is a serious problem if MBD is implemented completely in frequency domain .

$$\left(\mathbf{W}_{k_1}\mathbf{A}_{k_1} = \mathbf{P}_{k_1}\mathbf{\Lambda}_{k_1}\right) \neq \left(\mathbf{W}_{k_2}\mathbf{A}_{k_2} = \mathbf{P}_{k_2}\mathbf{\Lambda}_{k_2}\right)$$

Components on the same positions at different frequency bins do not belong to the same signal. Nonlinear function in frequency domain can be used as [7]

$$\Phi_k(Y_k) = \tanh\left(\eta \left| Y_k \right| \right) e^{j \arg(Y_k)}$$

7. H. Sawada, R. Mukai, S. Araki, S. Makino, "Polar Coordinate based Nonlinear Function for Frequence Domain Blind Source Separation," IEICE Trans. Fundamentals, Vol. E86-A, No. 3, March 2003.

Mixed implementation in time and frequency domain. Trade-off solution is to execute filtering in frequency domain and perform statistical independence test (that causes permutation indeterminacy) in time domain^{8,9}

 $\mathbf{X} = \mathrm{FFT}[\mathbf{x}]$ $\Phi_{i}(\mathbf{Y}_{i}) = \mathrm{FFT}[\boldsymbol{\varphi}(\mathbf{y}_{i})]$ $\mathbf{W}_{k}(l+1) = \mathbf{W}_{k}(l) + \mu [\mathbf{I} - \boldsymbol{\Phi}(\mathbf{Y}_{k})\mathbf{Y}_{k}^{H}]\mathbf{W}_{k}(l)$ $\mathbf{Y}_{k} = \mathbf{W}_{k}\mathbf{X}_{k}$ $\mathbf{y}_{i} = \mathrm{IFFT}(\mathbf{Y}_{i})$

Where k denotes frequency bin index, l denotes iteration index, i denotes signal index and μ is small learning gain.

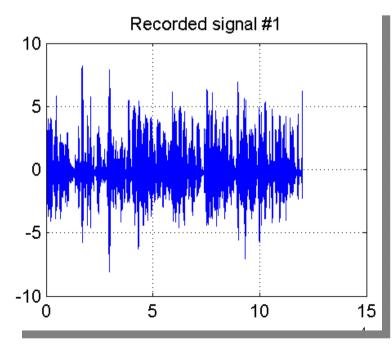
8 I.Kopriva, H. Szu, A.Persin, Optics Comm., Vol. 203 (3-6) pp. 197-211, 2002. 9. A. D. Back, A.C. Tsoi, Proc. of the 1994 IEEE Workshop – Neural Networks for Signal Processing IV, p.565, ed. 36/44 Vlontzos, J.N. Hwang, E. Wilson,

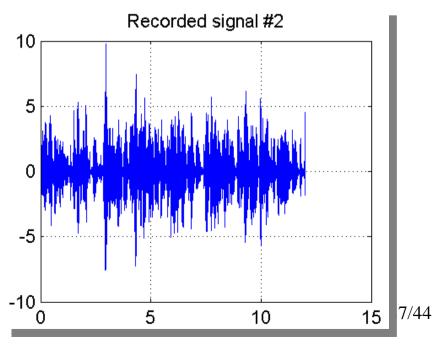
Applications of convolutive ICA

Speech separation in reverberant acoustic environment. Two recorded signals were downloaded from Russel Lamberts' home page:

http://home.socal.rr.com/russdsp/.

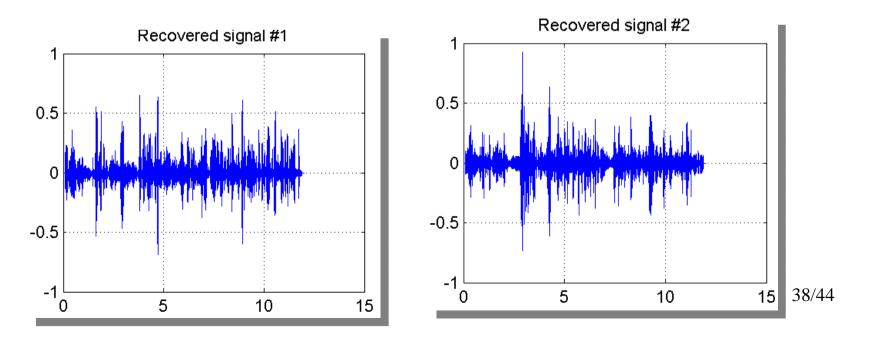
Signals were sampled with 8kHz and contain male and female speakers talking simultaneously for 12 seconds.





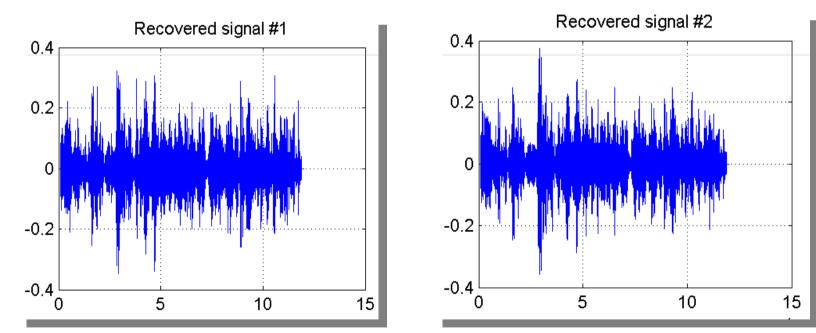
Applications of convolutive ICA

Parameters of the separation process were filter length *L*, Gaussian exponent α_i and learning gain μ .. At sampling frequency 8kHz and filter length *L*=1024 a relative delay of 64ms could be approximated. With speed of sound in the air of 330 ms⁻¹ this corresponds with path length difference of 21m. The following signals were recovered with *L*=1024 , α_i =1.0 and μ =0.005.



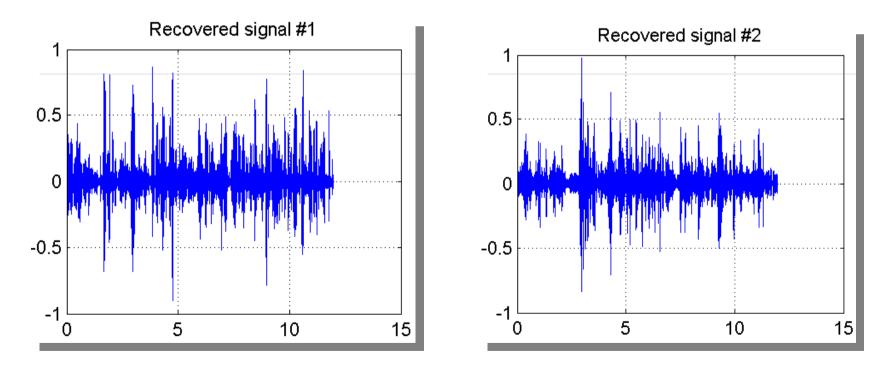
Applications of convolutive ICA

Following signals were recovered with *L*=1024 , α_i =3.0 and μ =0.005. Choice of α_i =3.0 corresponds with sub-Gaussian distributions and is wrong for speech signals.



Applications of convolutive ICA

Following signals were recovered with *L*=256, α_i =1.0 and μ =0.0013. Choice of *L*=256 will model relative path difference up to 5m.



If source signals are white (i.i.d.) **feedforward architecture can be employed in time domain** to realize on line non-causal implementation with natural gradient capable to approximate inverse of the NMP channels^{3,10}.

$$\mathbf{y}(t) = \sum_{p=0}^{L} \mathbf{W}_{p}(t) \mathbf{x}(t-p)$$
$$\mathbf{u}(t) = \sum_{p=0}^{L} \mathbf{W}_{L-p}^{H}(t) \mathbf{y}(t-p)$$
$$\mathbf{W}_{p}(t+1) = \mathbf{W}_{p}(t) + \eta(t) \left(\mathbf{W}_{p}(t) - \boldsymbol{\varphi}(\mathbf{y}(t-L))\mathbf{u}^{H}(t-p)\right) \quad p = 0, ..., L$$

where g() is functional inverse of φ such that:

$$\varphi_i(g_i(y_i)) = y_i$$

And *L* is filter length, *t* is time index, *p* is index of the coefficient of the matrix polynomial and η is small learning gain.

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10. S. Amari, S.C.Douglas, A. Cihocki and H.H. Yang, "Multichannel Blind Deconvolution and Equalization Using the Natural Gradient," IEEE International Workshop on Wireless Communication, Paris 1997, pp. 101-104.

Applications of convolutive ICA

Multichannel blind equalization of 3x3 systems with 3 i.i.d. source signals (QAM)¹⁰ with the NMP channels.

$$\mathbf{x}(k) = \sum_{i=1}^{7} \mathbf{A}_{i} \mathbf{x}(k-i) + \sum_{j=0}^{7} \mathbf{B}_{j} s(k-j), \quad (31)$$

$$\mathbf{A}_{1} = \begin{bmatrix} -0.56 + 0.35j & 0.34 + 0.08j & -0.14 - 0.40j \\ -0.28 + 0.08j & 0.18 + 0.43j & -0.66 - 0.15j \\ -0.08 - 0.27j & -0.40 + 0.15j & 0.16 - 0.36j \end{bmatrix} (32)$$

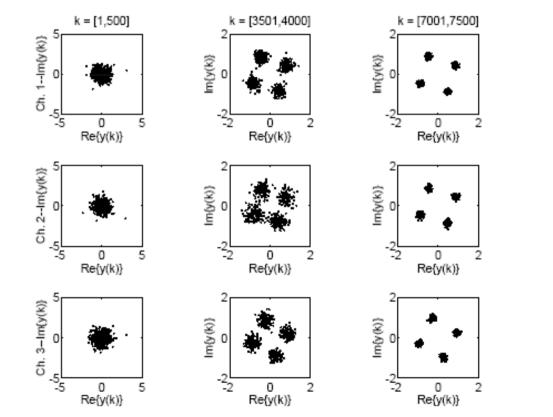
$$\mathbf{A}_{2} = \begin{bmatrix} 0.04 + 0.03j & 0.02 + 0.16j & 0.03 - 0.09j \\ 0.08 - 0.05j & 0.04 + 0.08j & -0.01j \\ 0.03 & 0.06 + 0.03j & 0.06 + 0.01j \end{bmatrix} \quad (33)$$

$$\mathbf{B}_{0} = \begin{bmatrix} 0.02 + 0.09j & 0.07 + 0.01j & 0.05 + 0.07j \\ 0.08j & 0.09 + 0.07j & 0.08 + 0.09j \\ 0.07 + 0.05j & 0.04 + 0.04j & 0.08j \end{bmatrix} \quad (34)$$

$$\mathbf{B}_{1} = \begin{bmatrix} 0.1 + 0.3j & 0.3j & 0.4 + j \\ 0.5 & 0.4 + 0.6j & 0.7 + 0.4j \\ 0.7 + 0.7j & 0.1 + 0.8j & 0.6 + 0.2j \end{bmatrix} . \quad (35)$$

Applications of convolutive ICA

Output constellations for blind equalizer are shown for all three restored sources for time intervals $1 \le t \le 500$, $3501 \le t \le 4000$ and $7001 \le t \le 7500$. $y_i(t) \approx s_i(t-4)$.



Applications of convolutive ICA

Output constellations are shown multichannel LMS equalizer trained with $d_i(t)=s_i(t-4)$. Constellations are shown on intervals $1 \le t \le 500$, $3501 \le t \le 4000$ and $7001 \le t \le 7500$.

