XVF3500 DSP Databrief

IN THIS DOCUMENT

1 Introduction

This document describes the voice DSP functionality available on the XVF3500 devices.

XVF3500 offers an integrated DSP solution which implements Acoustic Echo Cancellation (AEC), beamforming, dereverberation, noise suppression and gain control.

Supporting linear microphone arrays, XVF3500 provides up to 180° coverage and has been developed for far-field hands-free voice control in stereo TV and/or audio applications.

1.1 The Problem

High quality far-field communication with a device located several meters from the user requires a solution to the following problems:

▶ Poor speech-to-noise ratio (SpNR) due to the large distance (several meters) between the device and the user

▶ High reverberation of the speech signal due to the large separation between the device and user, and the room acoustics (with RT60 up to 0.8 seconds)

▶ High levels of acoustic echo due to the loudspeaker being located closer to the microphone than the user.

Figure 1 illustrates the above issues in a far-field communication system.

1.2 The Solution

The solution is to use advance DSP algorithms which deliver high speech-to-noise ratio, naturally sounding speech and eliminate acoustic echo while maintaining a transparent and low latency communication link.

The XVF3500 combines a four channel digital Pulse Density Modulation(PDM) microphone interface with the algorithms to deliver a high quality far-field communication experience. The key features of the solution are:

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Figure 1:
Far-field communication scenario
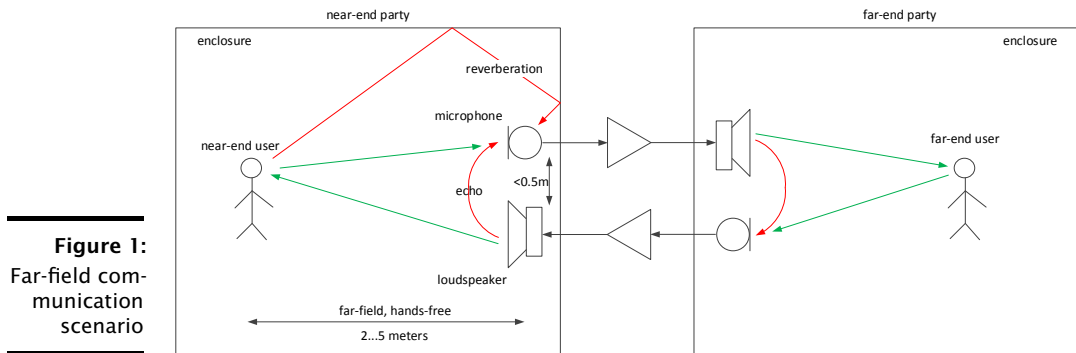
▶ High levels of Acoustic Echo Cancellation (AEC) and Suppression in living room conditions

▶ Stereo / multi-channel AEC

▶ State of the art, robust, and natural double-talk / full-duplex performance

▶ AEC Comfort noise insertion

▶ High speech clarity level even when users are at several meters distance, without requiring directional microphones

▶ Fast adaptive beamforming for tracking multiple near-end users

▶ Naturally sounding stationary / diffuse noise suppression

▶ Automatic gain control

A high level diagram of the solution is shown in Figure 2. The digital microphone signals are converted from PDM to Pulse Code Modulaton (PCM) format before being passed to the algorithms running on the XVF3500 device, along with the far-end signals that are played on the loudspeaker after having passed through a Digital to Analog Converter (DAC). An enhanced (mono) speech signal of the desired near-end speech without echo and reverberation is outputted along with an additional speech recognition signal.
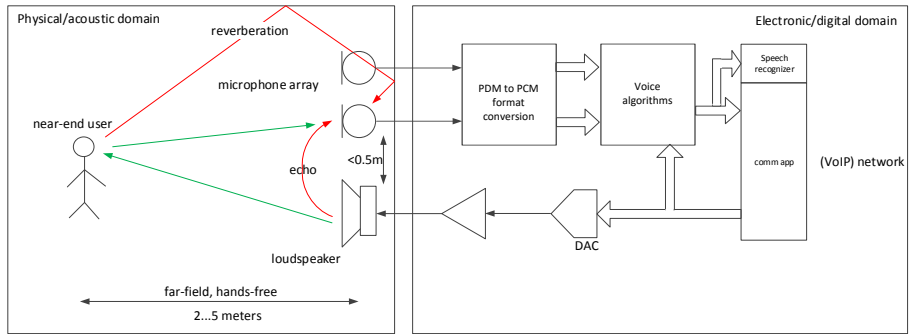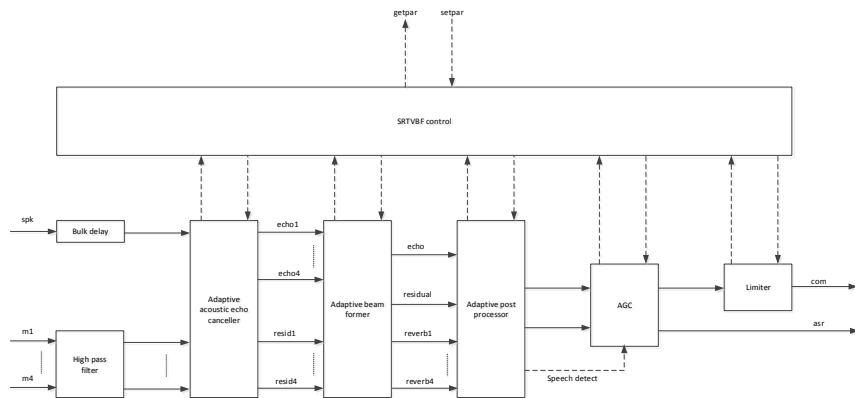
Figure 2:
Algorithm in
context

2   Functional Description

The processing pipeline consists of three main stages:

1. Adaptive Acoustic Echo Cancellation (AEC)

2. Adaptive Beamforming (BF)

3. Adaptive Spectral Post Processing.

A block diagram of the processing pipeline is shown in Figure 3. The four micro-
phone signals and the loudspeaker signals are inputs to the processing pipeline.
Two output signals are produced that contain the enhanced speech of the near-end
user: one optimized for human-to-human communication and the other optimized
for speech recognition engines.

Figure 3:
Block
diagram of
processing
pipeline

To help with speech intelligibility, a high-pass filter tuned is applied to the micro-
phone signals before passing them to the adaptive AEC.

The far-end signals are delayed by a bulk-delay to increase the efficiency of the AEC.

The AEC estimates the echo contribution in the microphone signals and subtracts the echo.

The beamformer filters the AEC residual signals and adapts its coefficients if near-end speech is detected. The beamformer outputs the enhanced signal via its primary output, provides reverberation reference signals, and provides a mono version of the echo signal, which are all passed to the post-processor.

For the communication output, the post-processor applies noise suppression to the primary output of the beamformer. Estimates of any remaining residual echo, stationary noise, and reverberation are used to perform spectral subtraction. The enhanced signal is then passed to Automatic Gain Control and Limiter components to ensure that the output signal has the desired signal strength.

For the speech recognition output, the post-processor applies less aggressive noise suppression and no echo suppression. The speech recognition output is then passed to Automatic Gain Control to ensure the output signal has the desired signal strength. No Limiter is applied to the speech recognition output.

The control component allows miscellaneous parameters to be set and the status queried, see section 3 for details of available parameters.

2.1   Adaptive Stereo Acoustic Echo Canceller

The Adaptive stereo AEC effectively suppresses echo generated when the loud-speaker outputs are received at the microphones. It is important that the stereo AEC is capable of removing the echo when loudspeakers and microphones are closely spaced in the same housing because the echo signal level may be significantly greater than the received near-end speech.

The stereo AEC is implemented with Frequency Domain Adaptive Filter (FDAF). The FDAF takes the delayed time-domain loudspeaker signal as input and generates the echo estimates and filter coefficients.

The echo estimates are subtracted from the filtered microphone signals to produce the residual signals which are used for adapting the filter coefficients of the FDAF. A block diagram for a single channel AEC with one far-end signal is shown in Figure 4 . The XVF3500 has two identical AEC blocks, one for each far-end signal.

Step-size control is used to prevent the filter coefficients from diverging excessively when the echo signal is smaller than the other contributions in the microphone signals i.e. near-end speech or noise. The step-size control also implements a continuous form of double-talk protection removing the need for a second echo path for double-talk detection.

The FDAF stops auto-adapting if the signal level falls below the user defined threshold (AECSILENCELEVEL) which by default is set to -80dBov (RMS).

Echo Return Loss (ERL) estimators are included and provide ERL estimates for two sub-bands. The sub-bands account for the non-flatness of a typical speech PSD
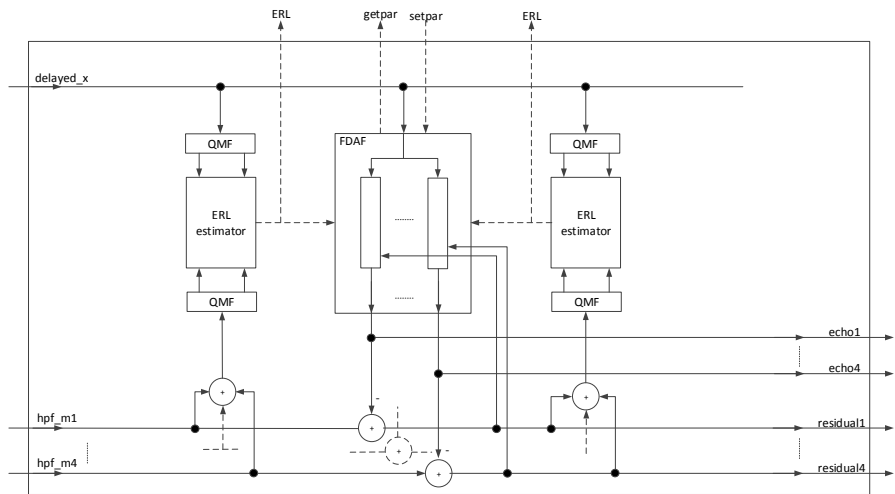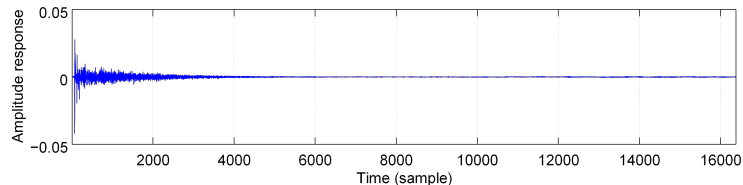
Figure 4:
AEC block
diagram

(Power Spectral Density), which leads (in combination with noise) to differences in convergence speeds across frequencies. The ERL estimates give an indication of the degree of convergence of the FDAF and of the quality of its echo estimates.

2.1.1   Echo Return Loss Enhancement

The length of the Adaptive Filter of the AEC has a direct influence on the Echo Return Loss Enhancement and therefore the overall performance of the AEC.

The adaptive filters are set to 2048 taps and the impulse response is shown in Figure 6.

The theoretically attainable ERLE with 2048 filter taps equals 13dB. To obtain sufficient echo reduction, the residual echo is further suppressed by the post-processor. Note that the adaptive filters assume a linear echo model and therefore do not remove any non-linear echo components; the latter task is performed by the post-processor.

Figure 5:
Adaptive
filter impulse
response
(reference)

Figure 6:
Adaptive
filter impulse
response,
2048 points
(linear arrays)
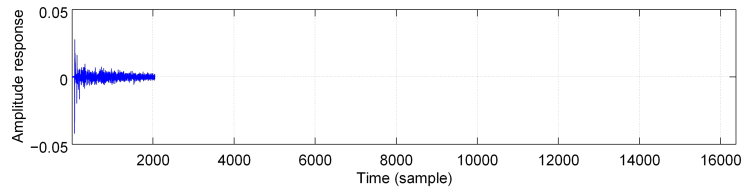
2.1.2 Acoustic Coupling

In far-field applications, the acoustic echo typically dominates the near-end signal observed at the microphones. A well designed device should have properly tuned acoustic coupling between the loudspeaker and microphones such that a 'loud' far-end signal does not result in clipping of the microphone signals.

Distortions in the digital microphone signals resulting from exceeding the maximum gain factor are not modelled by the AEC and thus should be prevented as much as possible to avoid excessive residual echo in the output of the algorithm. A less strict constraint which will in nearly all cases produce a system without problems of clipping the microphone signals in is to ensure that the magnitude of the transfer function between loudspeaker reference signal and each microphone signal should be less than 0dB. For example, the magnitudes of the equivalent transfer functions of the impulse response shown in Figure 6 are shown in Figure 7. As can be observed, the responses do not exceed 0dB anywhere.

Note that in practice the response will include reflections of the environment, and that the magnitude response will usually be smaller when measured in anechoic conditions. The best performance is achieved if the coupling is below 0dB at all frequencies, but not substantially lower than -20 dB.

### 2.1.3 Coefficient Norm Constraint

A certain amount of drift in the adaptive filter coefficients will occur due to a combination of finite-precision effects and the band-limited nature of the far-end signal due to DC removal and anti-aliasing filtering.

To avoid unchecked coefficient drift, a constraint is applied to the frequency domain coefficients such that they remain in the range [-`AECNORM`...+`AECNORM`].

2.2 Adaptive Beamformer

The Adaptive Beamformer focuses the microphones on the near-end source to perform noise suppression and other voice enhancements. An adaptive frequency domain Filtered Sum Beamformer (FSB) is used and takes reflections of the desired source into account.

The beamformer takes the AEC residuals as inputs and outputs a primary enhanced mono time-domain residual signal. The primary output signal will exhibit an

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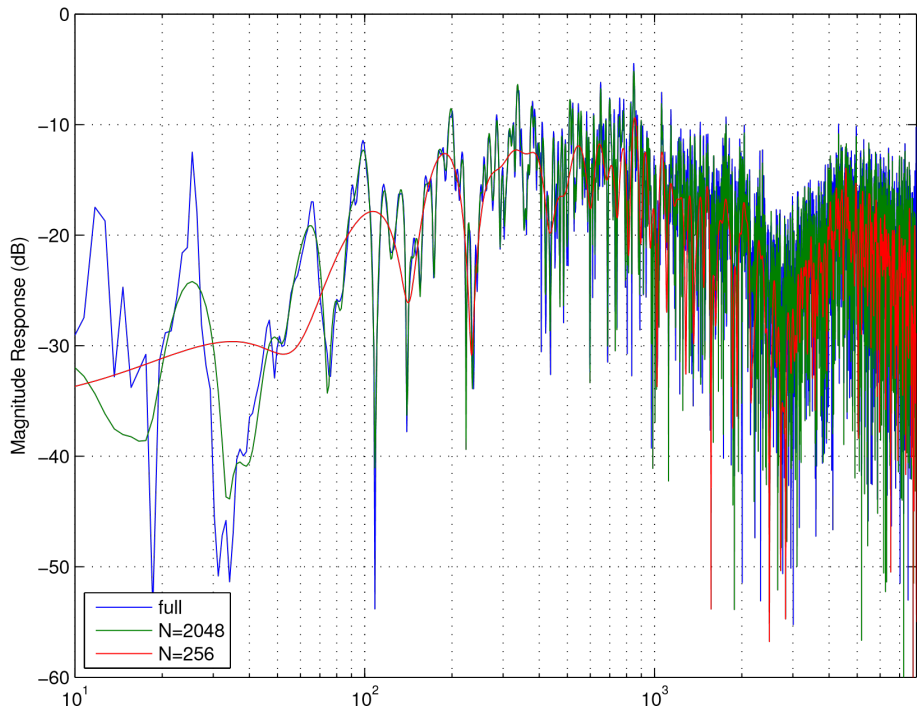
Figure 7: Magnitude of the transfer function of the example impulse responses

improved Clarity Index (CI) compared to the individual microphone signals and represents the first stage of the dereverberation process.
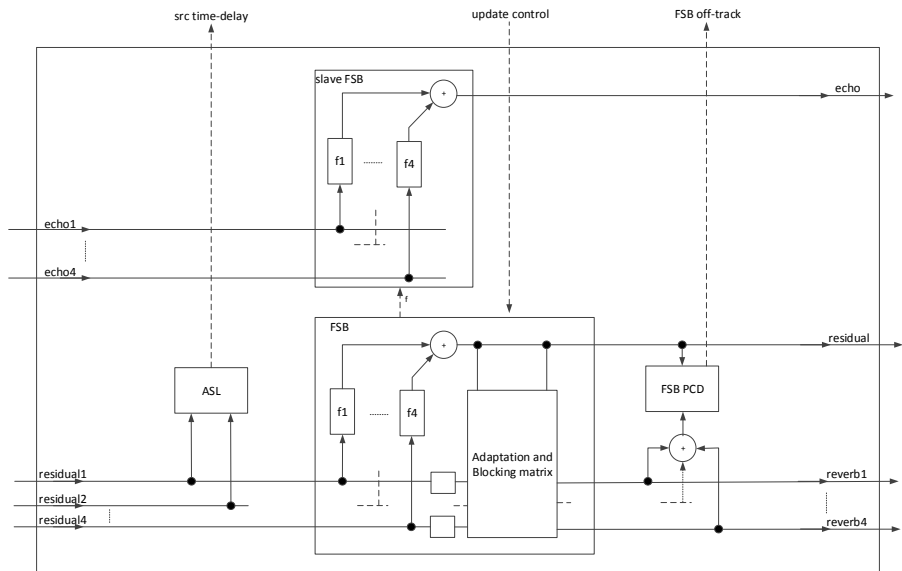
The beamformer adapts on desired near-end sources based on Audio Source Localization (ASL), and maximizes the power of its primary output under a constraint on filter coefficient magnitude.

A Path Change Detector (FSB PCD) triggers when a near-end source is detected at a significantly different location than the most recently active source. During these moments, the FSB is switched to a faster tracking mode.

A slave beamforming filter is used to construct the mono echo output signal; the slave filter coefficients are an identical (shallow) copy of the adaptive FSB coefficients.

2.2.1 Array Geometries

Larger array geometries typically result in a better clarity index improvement, as the clarity index of the beamformer depends on the both the number of microphones in the array as well as the spacing between microphones.

Figure 8:
Beamformer
block
diagram

The adaptive beamformer gives optimal attenuation of uncorrelated noise and adapts optimally in this type of noise. The power gain for the desired source in the beamformer output depends on the number of microphones (assuming omnidirectional elements), whereas the power gain for uncorrelated noise equals 1.

The amount of noise attenuation depends on the number of microphones in the array.

Reverberation is often modeled as a diffuse noise field and the coherence between two microphones in a diffuse noise field is a factor of the microphone spacing.

2.2.2   Audio Source Localization (ASL)

The Audio Source Localization method is based on a correlation method combined with peak picking. The time-delay/length-delay resolution is dependent on the sampling frequency. The best (theoretical) time-delay resolution of the algorithm is currently 8000×32 samples/s. Assuming the speed of sound is defined as $c = 344 m/s$, the length-delay quantization D corresponds to $344/(8000 \cdot 32) = 1.34375$ mm.

To relate the time-delay estimate to an angle between the array axis and the propagation vector of the sound field emanating from the sound source, the source is assumed to be located in the far-field and free-field. An illustration of a sound field passing over two microphones ($m1$, $m2$), resulting in a time-delay between the signals that are received at the microphones, is shown in Figure 9. The spacing

between the elements in this example is $d = 0.1m$. The sound field arrives at the microphones at an angular offset a. The resulting time-delay in the signals is measured and quantized to $n \cdot D$ meters ($n \in \mathbb{Z}$).
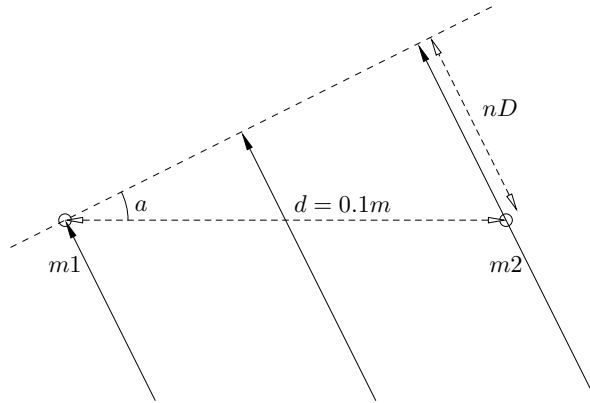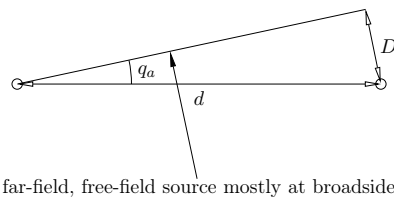

Figure 9:
Example of sound field passing over the array sensors

Note that the angle resolution (i.e. quantization) is dependent on the microphone spacing, and source location. When the source is approximately at broadside with respect to the array, the source angle is related to nD and microphone spacing d as

$$\sin(a) \approx \frac{nD}{d}. \tag{1}$$

The angle quantization q_a for e.g. a 10 cm array is then (Figure 10)

Figure 10:
Source localization resolution for broadside

far-field, free-field source mostly at broadside

$$q_a \approx \arcsin\left(\frac{D}{d}\right) \tag{2}$$

$$= \arcsin\left(\frac{0.00134375}{0.1}\right) \tag{3}$$

$$= 0.77° \tag{4}$$

If the source is located at approximately end-fire location, then the angle quantization is approximately (Figure 11)
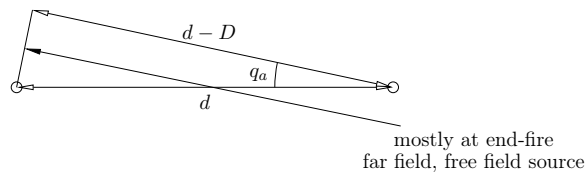
$$\cos(q_a) = \frac{d - D}{d}. \tag{5}$$

For example, for a 10 cm array:

$$q_a \approx \arccos\left(\frac{d - D}{d}\right) \tag{6}$$

$$= \arccos\left(\ \right) \tag{7}$$

$$= 9.4° \tag{8}$$

Figure 11:
Source localization resolution at end-fire

mostly at end-fire
far field, free field source

2.2.3 Beamformer Region-Of-Interest

The beamformer adapts on non-stationary sound sources that are within the defined 'Region-Of-Interest' (ROI).

The ROI is specified by the `BEAMWIDTH` and `BEAMANGLE` parameters which define a sub-range of the normalized range for which audio sources are defined as 'desired source.

The ROI is defined around the 'beamangle' direction, as measured with respect to the first two microphones in the array (Figure 12). Note that in terms of angular offsets, the ROI is not symmetrical around the specified `BEAMANGLE`.

The default values for `BEAMWIDTH` and `BEAMANGLE` are 0.5 and 0.0 respectively. Together they define a $60°$ region-of-interest as shown in the top left hand diagram in Figure 12.
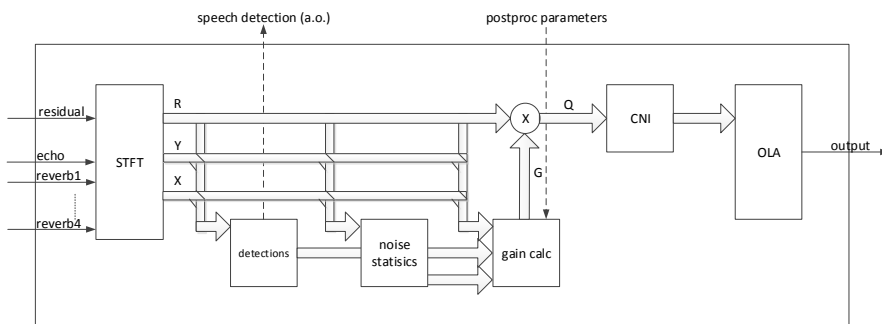
2.3 Post Processor

The Post Processor (Figure 13) uses spectral processing based on the Short Time Fourier Transform (STFT).

The time-domain input signals are converted to the frequency domain and the noise statistics are estimated. A spectral gain function is calculated and applied to the frequency domain residual signal.
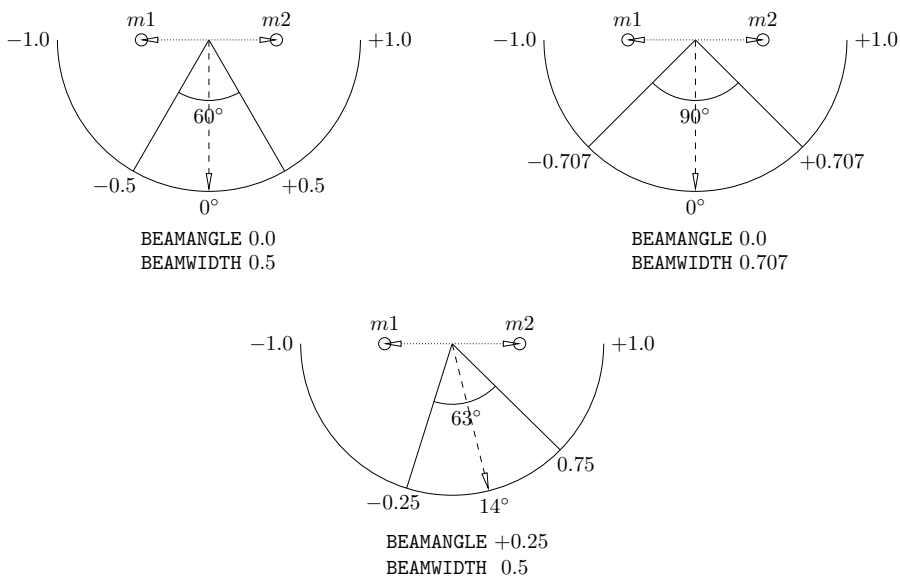
Figure 12:
Examples of ROI based on ASL for linear arrays

Figure 13:
Post Processor block diagram

Comfort Noise Insertion (CNI) is applied to the resulting signal to maintain a natural sounding output signal during suppression of residual acoustic echoes. The signal is then converted back to the time-domain by OverLap-Add (OLA) synthesis.

2.3.1 Stationary Noise

The magnitude response of the post-processor for stationary noise is a function of the estimated spectrum of the stationary noise and the spectrum of the residual.

The gain function always lies between 0 and 1, corresponding to attenuation factors in the range $[\infty...0]$ dB.

The user-configurable parameter GAMMA_NS allows overestimation of the noise estimate (leading to over-subtraction and larger attenuation), while parameter MIN_NS constrains in the maximum attenutation such that artefacts (e.g. musical tones) are reduced.

2.3.2 Non-Stationary Noise

The magnitude response of the post-processor for non-stationary noise is a function of the estimated spectrum of the non-stationary noise and the spectrum of the residual.

User configurable parameter, GAMMA_NN, allows over-estimation of the noise estimate (leading to over-subtraction, and larger attenuation), while parameter, MIN_NN constrains the maximum attenuation such that artefacts are reduced.

2.3.3 Residual Echo

The magnitude of the post-processor for residual echo noise is a function of the estimated spectrums of the residual direct/early linear and non-linear echo, and the echo tail.

The echo tail spectrum is recursively generated based on the current estimate and the early echo estimate, and is a factor of the estimated reverberation time T60.

If T60 estimate is too low, the tail will be underestimated and residual echoes will remain (unless user configurable parameter GAMMA_ETAIL is increased). If T60 is too high, the echo tail will be overestimated, and after post-processing no tail echoes will be heard, but at the cost of double talk performance.

T60 is estimated from the impulse response of the adaptive filter after the direct field. In systems with a large direct coupling and non-linearities it is difficult to estimate (the relative small coefficients) of the impulse response of the diffuse field accurately, leading to estimation errors in T60, with potential issues mentioned above.

2.4 HPF

The high pass filter on the microphone signals is implemented with a fourth order recursive filter. Three cut-off frequencies are supported: 70 Hz, 125 Hz, and 180 Hz. The responses of the microphone high pass filters are shown in Figure 14.

2.5 AGC

The AGC amplifies or attenuates its input signal such that the average power of the output signal has a certain desired level. The AGC adjusts its gain based on the power of the input signal during moments when voiced-speech is detected.
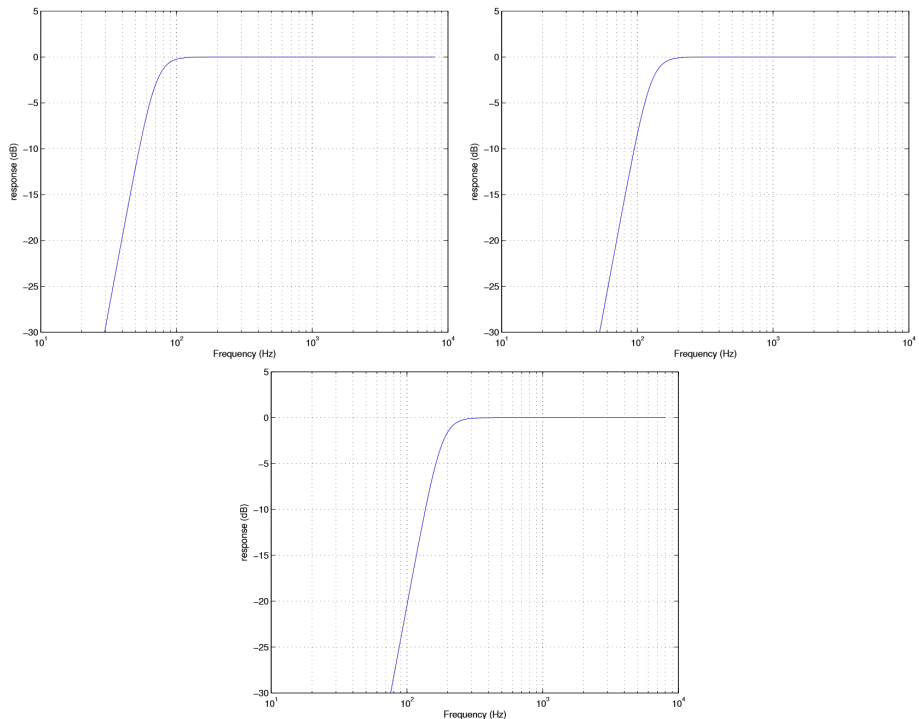
Figure 14:
Responses of
the
microphone
HPF

By default AGC is disabled (AGCONOFF is 0) and a fixed gain is applied to both output signals.

The initial gain at startup (AGCGAIN) is 32dB, and the time constant (AGCTIME) is set to 0.9 seconds. There is also a maximum gain (AGCMAXGAIN) which is set to 32dB by default to maintain a static gain level. This value limits the gain value to prevent that speech-like signals outside the room are amplified with a very high gain. When AGC is enabled, AGCDESIREDLEVEL defines the target power level.

2.6   Limiter

If exceptionally strong attacks in the output signal of the AGC occur, e.g. due to loud transient signals received by the microphones, the output signal might exceed the dynamic range supported by the receiver of the signal. To reduce the chance of a (hard) clipping operation, a limiter will activate if the power of the input signal exceeds a certain threshold. The default power threshold is -10.3dBov.

2.7   Speech Recognition Output

The speech recognition optimized output signal differs from the communication output in the following ways

▶ Limiter is not applied

▶ No comfort noise insertion

▶ No additional echo suppression (derverberation) in the post-processor

▶ Less aggressive noise suppression

3   XVF3500 DSP Parameters

The user configurable pre-processing parameters are listed in Table 1, the configurable AEC parameters are listed in Table 2 and the configurable Beamformer and Post Processor parameters are listed in Tables 3, 4 and 5.

Table 1:
Parameter
descriptions
for the pre-
processing

parameter	type	value	definition
MIC_ATTEN	APES_INT	[-100,0]	MIC input signal attenuator in decibels (read-write)
			Default: 0dB
AEC_REF_ATTEN	APES_INT	[-100,0]	AEC reference signal attenuator in decibels (read-write)
			Default: 0dB

parameter	type	value	definition
AECFREEZEONOFF	APES_INT	[0,1]	Adaptive Echo Canceler updates inhibit (read-write). 0 = Adaptation enabled (default) 1 = Freeze adaptation, filter only
AECNORM	APES_FLOAT1	[0.25 .. 16.0]	Limit of AEC filter coefficient values (read-write). Default: 2.0
AECSILENCELEVEL	APES_FLOAT1	[0.0 .. 1.0]	Threshold for signal detection in AEC (read-write). Equivalent range: $[-\infty .. 0]$ dBov Default: $1.0 \times 10^{-8} \Rightarrow -80$dBov $(10\log_{10}(1.0 \times 10^{-8}) \approx -80)$
AECSILENCEMODE	APES_INT	[0,1]	AEC far-end silence detection status (read-only). 0 = false (signal detected) 1 = true (silence detected)
HPFONOFF	APES_INT	[0..3]	High-pass Filter on microphone signals (read-write). 0 = OFF 1 = ON - 70 Hz cut-off (default) 2 = ON - 125 Hz cut-off 3 = ON - 180 Hz cut-off
RT60	APES_FLOAT1	[0.250 .. 0.900]	Current RT60 estimate in seconds (read-only).
RT60ONOFF	APES_INT	[0,1]	RT60 Estimation for AES (read-write). 0 = OFF 1 = ON (default)
AECERLMAX	APES_FLOAT1	[1.0 .. 99856.0]	Maximum erl estimate (write-only). Default: 99000.0
MAX_RT60	APES_FLOAT1	[0.0 .. 0.9]	Upper limit for the RT60 estimator in seconds (write-only). Default: 0.9
AEC_REF_DELAY	TYPE_INT	[0 .. 2400]	Parametric delay for the AEC reference samples. Value is given in samples at 16kHz. Default: 0

Table 2:
Parameter descriptions for the AEC module

parameter	type	value	definition
AGCDESIREDLEVEL	APES_FLOAT1	[0.0 .. 1.0]	Target power level of the output signal (read-write). Equivalent range: $[-\infty .. 0]$ dBov Default: $0.001 \Rightarrow -30$ dBov $(10\log_{10} 0.001 \approx -30)$
AGCGAIN	APES_FLOAT1	[1.0 .. 1000.0]	Current AGC gain factor (read-write). Equivalent range: $[0 .. 60]$ dB Default: $40.0 \Rightarrow 32$ dB $(20\log_{10} 40.0 \approx 32)$
AGCONOFF	APES_INT	[0,1]	Automatic Gain Control (read-write). 0 = OFF (default) 1 = ON
AGCMAXGAIN	APES_FLOAT1	[1.0 .. 1000.0]	Maximum AGC gain factor (read-write). Equivalent range: $[0 .. 60]$ dB Default: $40.0 \Rightarrow 32$ dB $(20\log_{10} 40.0 \approx 32)$
AGCTIME	APES_FLOAT1	[0.1 .. 1.0]	Ramp-up/down time-constant in seconds(read-write). Default: 0.9 s
CNIONOFF	APES_INT	[0,1]	Comfort Noise Insertion (read-write). 0 = OFF 1 = ON (default)
ECHOONOFF	APES_INT	[0,1]	Echo suppression (read-write). 0 = OFF 1 = ON (default)
FREEZEONOFF	APES_INT	[0,1]	Adaptive beamformer and postprocessor updates (read-write). 0 = Adaptation enabled (default) 1 = Freeze adaptation, filter only
GAMMA_NN	APES_FLOAT1	[0.0 .. 3.0]	Over-subtraction factor of non-stationary noise (read-write). min .. max attenuation (default: 1.1)
GAMMA_NS	APES_FLOAT1	[0.0 .. 3.0]	Over-subtraction factor of stationary noise (read-write). min .. max attenuation (default: 1.0)
MIN_NN	APES_FLOAT1	[0.0 .. 1.0]	Gain-floor for non-stationary noise suppression (read-write). Equivalent range: $[-\infty .. 0]$ dB Default: $0.3 \Rightarrow -10$ dB $(20\log_{10} 0.3 \approx -10)$
MIN_NS	APES_FLOAT1	[0.0 .. 1.0]	Gain-floor for stationary noise suppression (read-write). Equivalent range: $[-\infty .. 0]$ dB Default: $0.15 \Rightarrow -16$ dB $(20\log_{10} 0.15 \approx -16)$
NONSTATNOISEONOFF	APES_INT	[0,1]	Non-stationary noise suppression (read-write). 0 = OFF 1 = ON (default)

Table 3:
Parameter descriptions for the BF and PP

parameter	type	value	definition
STATNOISEONOFF	APES_INT	[0,1]	Stationary noise suppression (read-write). 0 = OFF 1 = ON (default)
TRANSIENTONOFF	APES_INT	[0,1]	Transient echo suppression (read-write). 0 = OFF 1 = ON (default)
FSBPATHCHANGE	APES_INT	[0,1]	FSB Path Change Detection (read-only). 0 = false (no path change detected) 1 = true (path change detected)
FSBUPDATED	APES_INT	[0,1]	FSB Update Decision (read-only). 0 = false (FSB was not updated) 1 = true (FSB was updated)
GAMMA_E	APES_FLOAT1	[0.0 .. 3.0]	Over-subtraction factor of echo (direct and early components) (read-write). min .. max attenuation (default: 1.0)
GAMMA_ENL	APES_FLOAT1	[0.0 .. 5.0]	Over-subtraction factor of non-linear echo (read-write). min .. max attenuation (default: 1.0)
GAMMA_ETAIL	APES_FLOAT1	[0.0 .. 3.0]	Over-subtraction factor of echo (tail components) (read-write). min .. max attenuation (default: 1.0)
NLAEC_MODE	APES_INT	[0..2]	Non-Linear AEC training mode (read-write). 0 = OFF (default) 1 = ON - phase 1 2 = ON - phase 2
NLATTENONOFF	APES_INT	[0,1]	Non-Linear echo attenuation (read-write). 0 = OFF (default) 1 = ON
VOICEACTIVITY	APES_INT	[0,1]	Signal energy exceeded a threshold (read-only). 0 = false (no voice activity) 1 = true (voice activity)
BEAMANGLE	APES_FLOAT1	[−1.0 .. 1.0]	Center of the beam for desired speech sources (read-write). Equivalent range: $[-90° .. 90°]$ Default: $0.0 \Rightarrow 0°$ $(\sin^{-1} 0.0 = 0°)$
BEAMWIDTH	APES_FLOAT1	[0.2 .. 1.0]	Width of the beam for desired speech sources (read-write). Equivalent range:$[23° .. 180°]$ Default: $0.5 \Rightarrow 60°$ $(2 \cdot \sin^{-1} 0.5 = 60°)$
DOAANGLE	APES_INT	[0 .. 180]	DOA angle; current value, orientation depends on build configuration (read-only).
FSBFREEZEONOFF	APES_INT	[0,1]	Adaptive beamformer updates (read-write). 0 = Adaptation enabled (default) 1 = Freeze adaptation, filter only

Table 4: Parameter descriptions for the BF and PP (continued)

parameter	type	value	definition
SPTHRESH	APES_FLOAT1	[0.0 .. 1.0]	Set parameter value for DNNS (read-write). Default: 0.0065
SR_ABSQFLOOR	APES_FLOAT	[0.0 .. 1000.0]	Absolute noise floor for voice activity detection (read-write). Equivalent range: $[-\infty .. 60]$ dB Default: $0.0 \Rightarrow -\infty$ dB $(20\log_{10} 0.0 = -\infty)$
SR_GAMMA_NN	APES_FLOAT1	[0.0 .. 3.0]	Gain-floor for non-stationary noise suppression (read-write). min .. max attenuation (default: 1.1)
SR_GAMMA_NS	APES_FLOAT1	[0.0 .. 3.0]	Over-subtraction factor of stationary noise (read-write). min .. max attenuation (default: 1.0)
SR_GAMMA_VAD	APES_FLOAT1	[0.0 .. 1000.0]	Threshold for voice activity detection (read-write). Equivalent range: $[-\infty .. 60]$ dB Default: $15 \Rightarrow 23.5$ dB $(20\log_{10} 15 \approx 23.5)$
SR_MIN_NN	APES_FLOAT1	[0.0 .. 1.0]	Gain-floor for non-stationary noise suppression (read-write). Equivalent range: $[-\infty .. 0]$ dB Default: $0.3 \Rightarrow -10$ dB $(20\log_{10} 0.3 \approx -10)$
SR_MIN_NS	APES_FLOAT1	[0.0 .. 1.0]	Gain-floor for stationary noise suppression (read-write). Equivalent range $[-\infty .. 0]$ dB Default: $0.15 \Rightarrow -16$ dB $(20\log_{10} 0.15 \approx -16)$
SR_NONSTATNOISEONOFF	APES_INT	[0,1]	Non-stationary noise suppression for ASR (read-write). 0 = OFF 1 = ON (default)
SR_STATNOISEONOFF	APES_INT	[0,1]	Stationary noise suppression for ASR (read-write). 0 = OFF 1 = ON (default)
XNLTRAINONOFF	APES_INT	[0,1]	Non-linear matrix training (read-write). 0 = OFF(default) 1 = ON

Table 5: Parameter descriptions for the BF and PP (continued)

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