Crosstalk channel estimation in xDSL systems based on SINR variations

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***Abstract*— Dynamic Spectrum Management (DSM) Level 2 allows to improve overall performance of multi-user Digital Subscriber Line (DSL) systems but relies on the a priori knowledge of interference (crosstalk) channels between twisted-pairs in a binder. While several crosstalk channel estimators have been proposed in the literature, they suffer from limitations that render them impractical in systems with legacy DSLs or require interruption of service during the training phase. In this work, a novel scheme is presented which overcomes these problems by identifying far-end crosstalk (FEXT) channels based on measured variations in the received signal-to-interference-plus-noise ratios (SINRs) due to individual transmitters switching on and off or adapting power spectra. Simulations show that an accuracy within a few dB can be obtained for dominant FEXT channels.**

I. Introduction

It is well-known that FEXT between copper wires in a telephone binder is the dominant impairment in current DSL systems [1], severely limiting achievable data rates. *DSM Level 2* [2] promises to improve overall system performance by centrally coordinating the modems transmit power allocation in order to minimize crosstalk between users. DSM techniques rely on Discrete Multitone (DMT) transmission, which is a multi-carrier scheme equivalent to Orthogonal Frequency Division Multiplex (OFDM) and allows to adapt the bit- and power-loading for each narrow-band subcarrier (*tone*) to the actual channel conditions. Furthermore, in order to compute optimal transmit spectra, a Spectrum Management Center (SMC) requires complete knowledge of the multi-user interference channel, i.e. the frequency-selective characteristics of all direct and crosstalk channels. However, since in DSM Level 2 transmit power levels are optimized, only the magnitude of the generally complex valued channel response has to be available. Therefore, in an N-user DSM system, for each tone, N direct channel and $N(N-1)$ FEXT channel gain coeffients have to be estimated.

While the current xDSL standards [3] already provide estimates of the direct channel for different subcarrier groups, crosstalk channel gain coefficients are not available. To avoid the necessity for modifications to existing Customer Premises Equipment (CPE), it is desirable that techniques to acquire channel knowledge utilize only measurements available by current standard-compliant hardware. In [4], the authors propose an estimator based on measurements obtained in Loop Diagnostic mode. However, the described scheme involves muting individual users during the training phase and as such implies a forced interruption of service for users that have already started a DSL session, i.e. entered “showtime”. In order not to impact customer satisfaction, service providers avoid such measures which is why such a scheme is not practical. Another approach [5] is based on adding training sequences superimposed onto the actual data signals and as such is only practical for downstream transmission.

In this work, a novel scheme for FEXT channel estimation is proposed which relies entirely on measurements available in current standard-compliant systems, but, in contrast to prior work, avoids the need to interrupt transmission of users during the estimation phase. The algorithm exploits the measured variation of SINR at each receiver in the system that occurs due to the fluctuation of crosstalk some user causes by initiating or terminating a session, or due to a modem performing online adaption of power levels. The latter includes adjustment of fine gains to equalize the signal-to-noise ratio (SNR) margin across tones, as well as the low-power mode currently under discussion for adoption in future systems. Given enough SINR measurements at different time instances where the crosstalk profile has changed significantly, FEXT magnitude coefficients can be estimated uniquely without additional knowledge. Following a Recursive Least-Squares (RLS) approach, the estimates can be refined as new measurements arrive. The scheme also allows to identify the missing FEXT coefficients once a user joins the DSM system and is equally suited for estimation of up- and downstream channels. The remainder of this work is structured as follows: in Section II, the system model underlying our proposed estimator is defined. Section III outlines the principles of the novel approach which for which a recursive updating scheme is presented in Section IV. Finally, Section V discusses some performance results obtained from numerical simulations.

II. System Model

Consider an N-user DSL channel with loops $n = 1, \dots, N$ sharing the same binder, thus causing mutual FEXT on each other's line. By employing DMT transmission with K orthogonal tones $k = 1, \dots, K$, the interference channel is effectively divided into K independent subchannels k. Augmenting each DMT symbol with a sufficiently long cyclic extension allows the direct channel of line n for tone k to be fully described by a single complex coefficient $h_k^{n,n}$. Similarly, the crosstalk channel from disturber m to victim line n on tone k is given by the complex scalar $h_k^{n,m}$ ($m \neq n$). The channel coefficients are assumed to be time-invariant.

Let $s_k^n(t)$ denote the transmit power spectral density (PSD) of line n on tone k at time instance t and let $\sigma_k^n(t)$ denote the external noise PSD comprising all disturbances other than the crosstalk from the $N-1$ disturber lines. Note that the case of line n not being active at time t is modeled by setting $s_k^n(t) = 0$ ($\forall k = 1, \dots, K$). The received noise PSD $w_k^n(t)$ and SINR $\gamma_k^n(t)$ are then given by

$$w_k^n(t) = \sum_{m \neq n} |h_k^{n,m}|^2 s_k^m(t) + \sigma_k^n(t) \tag{1}$$

and

$$\gamma_k^n(t) = \frac{|h_k^{n,n}|^2 s_k^n(t)}{w_k^n(t)}, \tag{2}$$

respectively.

III. FEXT channel estimation

The main idea of the proposed estimator is to combine a sufficient number of measurements $\hat{w}_k^n(t)$ of noise powers $w_k^n(t)$ from different time instances $t \in \{1, \dots, U\}$ where disturber lines have switched on or off or adapted their transmit PSD in order to obtain a unique estimate of the crosstalk gain coefficients $g_k^{n,m} = |h_k^{n,m}|^2$. For this to work out, the external noise power $\sigma_k^n(t)$ is assumed to be constant, i.e. $\sigma_k^n(1) = \dots = \sigma_k^n(U) = \sigma_k^n$.

In current DSL systems, the values $\hat{w}_k^n(t)$ are either obtained from quiet line noise (QLN) values measured in a double-ended line test (DELT) at the initialization of a new session, or, if the modem is already in showtime, derived from estimates $\hat{g}_k^{n,n}(t)$ and $\hat{\gamma}_k^n(t)$ of the direct channel gain $g_k^{n,n} = |h_k^{n,n}|^2$ and SINR γ_k^n along with the current power levels $s_k^n(t)$ according to

$$\hat{w}_k^n(t) = \frac{s_k^n(t)\hat{g}_k^n(t)}{\hat{\gamma}_k^n(t)}. \tag{3}$$

Note that, due to the nonlinear transformation of $\hat{\gamma}_k^n(t)$, $\hat{w}_k^n(t)$ is generally biased even if $\hat{\gamma}_k^n(t)$ is unbiased and the bias grows as the variance $\mathrm{Var}\{\hat{\gamma}_k^n(t)\}$ grows. To not overly complicate the estimation problem, however, we accept a possible biasedness of $\hat{w}_k^n(t)$ in the scope of this work.

Given a sequence $\hat{\mathbf{w}} = \{\hat{w}_k^n(1), \dots, \hat{w}_k^n(U)\}$ of noise power estimates, we obtain the linear observation model for tone k of line n according to eq. (4) shown at the top of the following page, where $\mathbf{e}$ is an error term comprising measurement, estimation and quantization errors in the values of $\hat{g}_k^{n,n}$ and $\hat{\gamma}_k^n$.As the statistics of $\mathbf{e}$ depend on the proprietary estimation algorithms and hardware specifications of the DSL modems, we abstain from explicitly assuming a statistical model for $\mathbf{e}$. Rather than that, we choose a least-squares (LS) approach where the estimate $\hat{\boldsymbol{\theta}}$ of the parameter vector $\boldsymbol{\theta}$ as defined in eq. (4) is chosen according to

$$\begin{aligned} \hat{\boldsymbol{\theta}} &= \arg\min_{\boldsymbol{\theta}} \|\hat{\mathbf{w}} - \mathbf{S}\,\boldsymbol{\theta}\| \\ &= (\mathbf{S}^T\mathbf{S})^{-1}\mathbf{S}^T\hat{\mathbf{w}}. \end{aligned} \tag{5}$$

The LS solution $\hat{\boldsymbol{\theta}}$ exists and is unique if and only if the columns of $\mathbf{S}$ are linearly independent [6]. This implies that in order to obtain a unique estimate, we require at least $N+1$ measurements, i.e. $U \geq N+1$, as well as that the power allocation $s_k^n(t)$ of each user n is not constant for all time instances $t = 1, \dots, U$. Fortunately, this is unlikely if the modems perform power adaption like fine gain adjustment.

IV. Recursive estimation

The estimates obtained by eq. (5) can be refined as new noise observations $\hat{w}_k^n(t)$ are acquired at subsequent time instances. To best improve accuracy of the estimates, it is desirable that the new measurements are obtained after power allocation of

$$\underbrace{\begin{pmatrix} \hat{w}_k^n(1) \\ \vdots \\ \hat{w}_k^n(U) \end{pmatrix}}_{\hat{\mathbf{w}}} = \underbrace{\begin{pmatrix} 1 & s_k^1(1) & \cdots & s_k^{n-1}(1) & s_k^{n+1}(1) & \cdots & s_k^N(1) \\ \vdots & & \vdots & \vdots & & \vdots & \vdots \\ 1 & s_k^1(U) & \cdots & s_k^{n-1}(U) & s_k^{n+1}(U) & \cdots & s_k^N(U) \end{pmatrix}}_{\mathbf{S}} \underbrace{\begin{pmatrix} \sigma_k^n \\ g_k^{n,1} \\ \vdots \\ g_k^{n,n-1} \\ g_k^{n,n+1} \\ \vdots \\ g_k^{n,N} \end{pmatrix}}_{\boldsymbol{\theta}} + \mathbf{e} \tag{4}$$

one or several transmitters has changed significantly. At time instance t, when a new measurement

$$\hat{w}_k^n(t) = \sum_{m \neq n} s_k^m(t) g_k^{n,m} + \sigma_k^n + e_k^n(t) \tag{6}$$

has become available, $\hat{\mathbf{w}}$ and $\mathbf{S}$ are augmented according to

$$\hat{\mathbf{w}}(t) = \begin{pmatrix} \hat{\mathbf{w}}(t-1) \\ \hat{w}_k^n(t) \end{pmatrix} \quad \text{and} \tag{7}$$

$$\mathbf{S}(t) = \begin{pmatrix} \mathbf{S}(t-1) \\ \mathbf{s}^T(t) \end{pmatrix} \tag{8}$$

where $\mathbf{s}(t+1)$ is a column vector with elements $s_k^n(t+1)$ arranged in the same way as the rows of $\mathbf{S}$ in (4). Rather than solving the LS problem (5) anew at each time step t, a RLS approach [6] allows to update the old estimate $\hat{\boldsymbol{\theta}}(t-1)$ from previous time step $t-1$, yielding a new $\hat{\boldsymbol{\theta}}(t)$ that is again optimal in the LS sense. The recursion equations are summarized as follows:

$$\begin{aligned} \hat{\boldsymbol{\theta}}(t) &= \hat{\boldsymbol{\theta}}(t-1) + \mathbf{k}(t)\Big[\hat{w}_k^n(t) - \mathbf{s}^T(t)\hat{\boldsymbol{\theta}}(t-1)\Big] \\ \mathbf{k}(t) &= \gamma(t)\mathbf{P}(t-1)\mathbf{s}(t) \\ \gamma(t) &= 1 + \mathbf{s}^T(t)\mathbf{P}(t-1)\mathbf{s}(t) \\ \mathbf{P}(t) &= \mathbf{S}^T(t)\mathbf{S}(t) \\ &= \mathbf{P}(t-1)\Big[1 - \gamma(t)\mathbf{s}^T(t)\mathbf{s}(t)\mathbf{P}(t-1)\Big]. \end{aligned} \tag{9}$$

While the batch LS method, i.e. solving the problem non-recursively via eq. (5), has a complexity $\mathcal{O}(U^3)$ [7], the overall complexity of the recursion (9) is given by $\mathcal{O}(U^2)$ so that the reduction in computational cost increases as the number U of measurements grows.

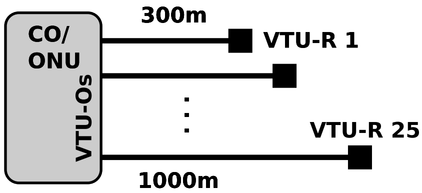


Fig. 1: CO-based deployment scenario

V. Performance

To evaluate performance of the described FEXT estimation scheme, numerical simulations were carried out in a downstream scenario shown in Figure V. 25 VDSL2 loops are deployed from the Central Office (CO) with loop lengths equally spaced between $300\,\mathrm{m}$ and $1000\,\mathrm{m}$. The VTU-Os are transmitting with a flat PSD of $-60\,\mathrm{dBm/Hz}$ within a bandwidth of $17.7\,\mathrm{MHz}$. The binder is of TP100 type [8] and the FEXT channels are generated according to the NIPP-NAI MIMO model [9]. In addition to FEXT, each receiver input is subject to white background noise with PSD $-140\,\mathrm{dBm/Hz}$ [8].

Accuracy of the novel estimator has been analyzed for FEXT channels from three disturber lines n_A, n_B and n_C to the victim line n_V with loop length $700\,\mathrm{m}$. Figure 2 shows the generated channel gains of all 24 disturbers of n_V. The crosstalk channels of disturbers n_A, n_B and n_C are drawn with solid lines where the strongest channel corresponds to disturber n_A and the weakest channel to disturber n_C. The crosstalk coupling strength varies in a range of roughly $27\,\mathrm{dB}$ between channels which is due to the different disturber loop lengths as well as other factors like geometric distance of twisted pairs in a binder as modeled by the NIPP-NAI coupling matrix [9].

For the estimation, $U = 50$ SNR measurements $\hat{\gamma}_k^{n_V}(t)$ and direct channel gain estimates $\hat{g}_k^n(t)$ per tone k are acquired at time steps $t = 1, \ldots, U$. Activity of individual disturber loops is determined

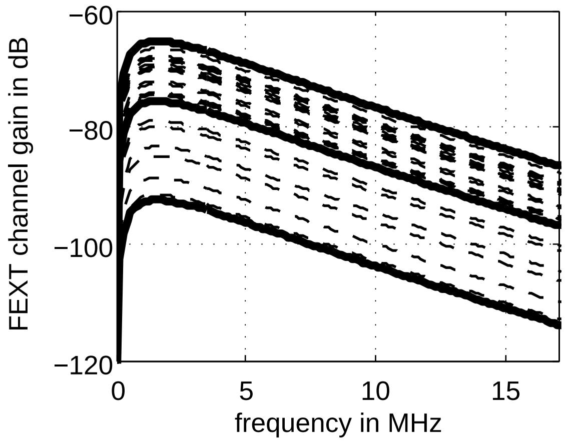


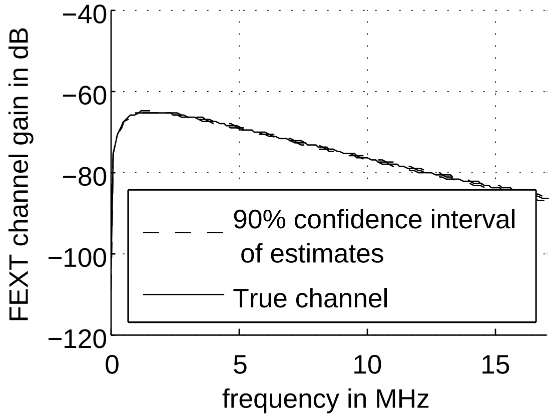
Fig. 2: 24 FEXT channels of victim line n_V. The channels of disturbers n_A, n_B and n_C are drawn with solid lines.

randomly at each time instance: a loop n is active at time t with probability 0.4, otherwise $s_k^n(t) = 0$.

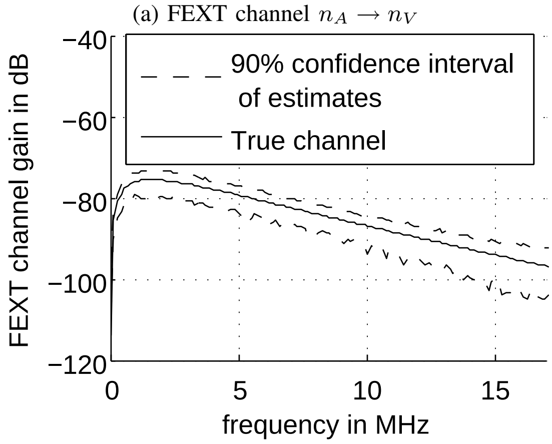
To simulate measurement and estimation errors in both quantities, channel gain and noise estimation algorithms from [1] have been implemented to generate the measured values. Gain estimation uses a training sequence of 1000 symbols with constant magnitude and uniformly distributed phase. Simulation of noise estimation assumes correct decisions of the QAM decoder and averages over 1000 received symbols. The crosstalk plus background noise signal on each tone is modeled as white complex Gaussian noise.

Figure 3 shows the 90% confidence intervals of the magnitude estimates of the three crosstalk channels obtained from 300 Monte-Carlo runs.

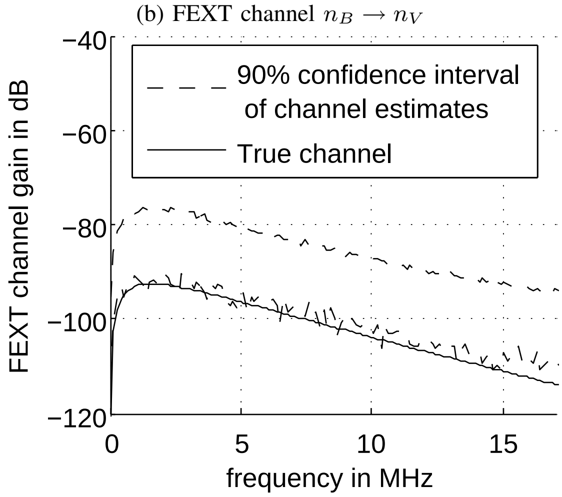
As one could have expected, estimation error variance is highly dependent on the strength of the crosstalk channel. While the estimates of the weakest channel $n_C \rightarrow n_V$ are subject to high uncertainty in addition to a significant bias, gain coefficients for channel $n_A \rightarrow n_V$ are acquired with great accuracy of roughly 1 dB. Obviously, the received crosstalk signal from disturber $n_C \rightarrow n_V$ is over-powered by the other stronger FEXT channels as well as the background noise so that an accurate estimate is not possible here. However, this is not critical because the such weak crosstalk channels are minor relevance in DSM spectrum optimization. The bias can be explained by the bias inherent to the noise power estimates obtained via (3) which becomes more pronounced in the low SNR regime.



(a) FEXT channel $n_A \rightarrow n_V$



(b) FEXT channel $n_B \rightarrow n_V$



(c) FEXT channel $n_C \rightarrow n_V$

Fig. 3: 90% confidence intervals for estimates of three different FEXT channels

VI. Conclusion

In this work, a novel scheme for online acquisition of FEXT channel gain coefficients in DSL binders relying only measurements available in current systems has been presented. The algorithm overcomes most limitations of previous techniques like service interruption of lines during the estimation phase and allows accurate estimation of the dominant disturber channels which are also the most relevant in DSM systems. Using a recursive update scheme, the computational cost can be significantly reduced.

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