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## An improved channel model for ADSL and VDSL systems

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### Abstract

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### Keywords

digital filters, digital subscriber lines, impulse noise, transmission line theory

### Disciplines

Physical Sciences and Mathematics

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# An Improved Channel Model for ADSL and VDSL Systems

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## Abstract

This paper examines existing channel models used with xDSL systems and identifies a key shortcoming - namely, the implicit assumption that all impulse noise originates at the transmitter. Based on extensive data collected from the local loop, a new model is proposed which addresses this problem by combining a digital filter model of the transmission line with a distributed noise source. This better reflects the nature of a real telephone line, and thus provides a more solid basis for simulation and optimisation of xDSL systems.

## 1. Introduction

The voracious demand for fast domestic Internet access has created the need for new technologies for delivering improved data rates at ever-decreasing prices. A critical component of any such high-bandwidth network systems is the final element of the network, connecting the end-user to the network. Although this part of the network is usually relatively short, it is by far the most expensive.

Several technologies are now starting to emerge as viable solutions to this so-called 'last mile' problem, including Hybrid Fibre/Coaxial (HFC) networks, various wireless systems and the Digital Subscriber Line (DSL) family. The latter technology enjoys the advantage of being largely compatible with the existing voice telephony network. Therefore, DSL technologies should offer an attractive medium-term solution, particularly where extensive existing copper telephone networks already exist.

Two of the key parameters which determines the effectiveness of xDSL systems are the frequency response and the noise characteristics of the copper telephone line. A useful channel model should therefore accurately represent both aspects of the system. A channel model is proposed here which combines both the frequency response characteristics of an ideal telephone line with an impulse noise model in which the physical point of ingress may be anywhere on the line.

## 2. ADSL Fundamentals

The two main DSL technologies to be discussed in this paper are the Asymmetric Digital Subscriber Line (ADSL) and Very high-speed Digital Subscriber Line (VDSL) systems. ADSL was standardised in 1995 [4] (updated in 1998 and 1999), while the

VDSL standard is still being developed (a draft standard is available from the VDSL Coalition [2]).

The environment in which xDSL technologies operate is relatively harsh, with unshielded copper telephone lines suffering from a variety of noise impairments, as well as a complicated frequency response which varies from line to line. ADSL attempts to mitigate both of these problems via a flexible modulation scheme, powerful coding techniques, and relatively simple channel equalisation.

The modulation scheme used in ADSL is a multiple carrier system known as Discrete Multi-Tone modulation (DMT), in which different parts of the spectrum may be allocated a different bits per symbol loading, tailored to suit the particular Signal-to-Noise Ratio (SNR) in that part of the spectrum. This allows narrow-band interferers to be notched out of the spectrum, with the added benefit of increasing the symbol period, thereby spreading the effect of impulse noise out over many samples. It is, however, noted by Akansu *et al* [1] that where narrow-band interferers are not precisely centred on one of the DMT sub-carrier frequencies, significant Inter-Symbol Interference (ISI) will occur due to leakage.

ADSL uses a Forward Error Correction (FEC) scheme based on Reed-Solomon coding and (optionally) 16-state 4-dimensional trellis coding. Incoming data streams, which may include up to four simplex and three duplex streams, are either directed to a fast buffer (with  $> 2$  ms delay) or a slower interleaved buffer ( $> 20$  ms delay) which is more resilient to impulse noise errors.

The equaliser used in ADSL systems is a simple one tap / sub-channel scheme. This approach seems to work well enough over standard telephone lines, since within one sub-channel the phase distortion and attenuation do not usually vary greatly.

Despite the noise mitigation and error correction strategies implemented in xDSL systems, the BER is still quite high compared to network media such as Ethernet or fibre networks - even when the interleaved buffer is used. As with wireless networks, such media errors can seriously degrade throughput. Much of the problem is due to the fact that standard TCP assumes that dropped packets are due to congestion rather than noise, and hence it will temporarily choke off throughput when packets are lost.

## 3. Existing Channel Models

A good understanding of the channel over which xDSL systems are to operate is essential in obtaining the best possible channel

capacity. This requires consideration of both the frequency response characteristics and the noise conditions present on these lines. Although several models for communications channels exist ([3], [5]), each makes a number of simplifying assumptions which may reduce the accuracy of the model when applied to telephone lines. One such simplification is the implicit assumption that all impulse noise events have their point of origin at the transmitter.

Most channel models separate the noise sources from the frequency response of the channel. A noise 'excitation' waveform is generated and passed through a simulated noiseless channel, in the form of a digital filter. One may then modulate a data stream as described in the ADSL standard (or with one of the proposed VDSL modulation schemes), add the desired amount and type of noise, feed it through the simulated ideal channel, and observe the effects.

The statistics traditionally used to model the probability density function and waveform of the sum of the two noise types is relatively simple. Each sample of the unfiltered noise is a sum of two random variables representing the noise. One random variable represents Additive White Gaussian Noise (AWGN) with a mean of zero and a variance  $\sigma_g^2$ , while the other is the product of a simple Bernoulli distribution and a second distribution which represents the amplitudes of the impulses. Several models have been proposed for the impulse amplitude distribution, including a Gaussian distribution [3], with a mean of zero and a variance of  $\sigma_i^2$ , and the so-called Stable distribution [5].

The resultant time-domain signal may be described as:

$$y(k) = G_1 x(k) + G_2 g_1(k) + G_3 b(k)g_2(k) \quad (1)$$

where  $G_1$  is a signal attenuation factor,  $G_2$  is the gain term for the Gaussian noise  $g_1(k)$ ,  $G_3$  is the gain term for the impulsive noise ( $b(k)$  is the Bernoulli random variable, while  $g_2(k)$  is the Gaussian or stable-distributed random variable corresponding to the impulse amplitudes).

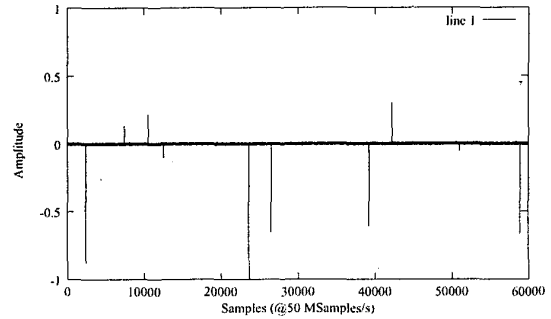
In existing models, then, Equation 1 may be considered the excitation to the system, which is then filtered by a noiseless channel. An example waveform for this excitation signal is shown in Figure 1(a).

Given that the noise signal (however it is generated) exists, we now consider the noiseless filter model of a telephone line. Typically, a telephone line consists a series of segments of unshielded twisted pair cable, with each segment possessing a different (and finite) distributed capacitance, resistance and inductance (as suggested in [9] and [8]). The equivalent circuit model is shown in Figure 2.

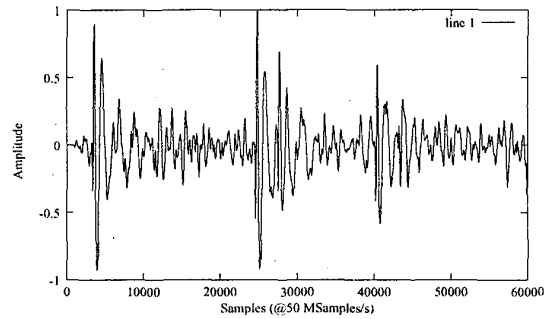
An expression for the transfer function of this system may be found by the recursive relation given by

$$H(s) = \prod_{k=1}^N \frac{N_{Rek}(s)}{D_{Rek}(s)(Ls + R) + N_{Rek}(s)} \quad (2)$$

where



(a) Excitation



(b) Received waveform

Figure 1: Noise: before and after filtering through noiseless channel

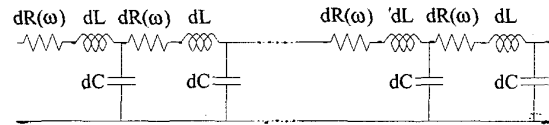


Figure 2: Distributed RLC model for telephone line.

$$Z_{Rek}(s) = \frac{N_{Rek}(s)}{D_{Rek}(s)} = \frac{(Ls + R)D_{Re(k-1)}(s) + N_{Re(k-1)}(s)}{s(LCs + RC)D_{Re(k-1)}(s) + D_{Re(k-1)}(s) + CsN_{Re(k-1)}(s)} \quad (3)$$

and

$$Z_{Re0} = \begin{cases} \frac{N_{end}}{sCN_{end} + D_{end}} & \text{(general termination)} \\ \frac{1}{sC} & \text{(open-circuit termination)} \end{cases} \quad (4)$$

Due to roundoff error, the above model is unwieldy and inaccurate unless arbitrary-precision floating point numbers are used.

An alternative approach, which has the added advantage of being able to easily account for the skin effect, is to use nodal analysis to calculate the frequency response of the circuit at a certain number of frequencies. This may then be used as the basis for a digital filter model (by the methods outlined in [7]).

A signal generated by filtering the waveform in Figure 1(a) via this approach is shown in Figure 1(b).

#### 4. System Identification

To verify the circuit model, it is necessary to compare the impulse response of the simulated line with practical measurements. A large set of impulse noise data gathered from a number of different telephone lines has been provided by Telstra Corporation<sup>1</sup>. After examining many of these waveforms it is clear that individual impulse events as seen by the receiver are of widely varying scales.

Despite this variation, it is possible to estimate the impulse response of the whole channel using a method similar to that used by Li, Huang, Ingram and Howard [6] for HFC networks. This technique involves using a Singular Value Decomposition (SVD) to reduce a set of impulse noise waveforms to a small number of basis functions.

Firstly the sampled waveforms are lined up in time and normalised. The SVD is calculated, which generates a set of basis functions and a measure of how well each basis function fits the overall set of data. The 'optimal impulse' calculated from the Telstra data is shown in Figure 4.

To verify the result, the normalised cross-correlation coefficient between this optimal impulse and each of the sample waveforms was calculated. In the majority of cases, this was found to be very close to 1.

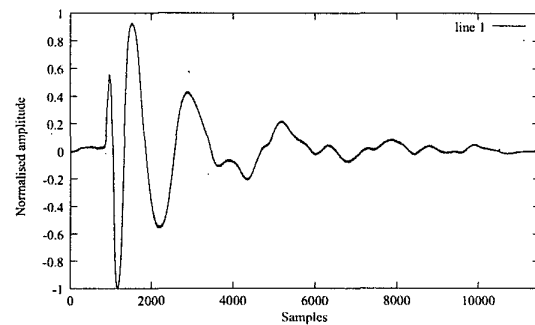
System identification by application of Prony's method to the optimal impulse resulted in a filter with 35 poles and 30 zeros. The Mean Squared Error between this approximate system and the optimal impulse was  $1.9 \times 10^{-6}$ .

#### 5. Unified Model

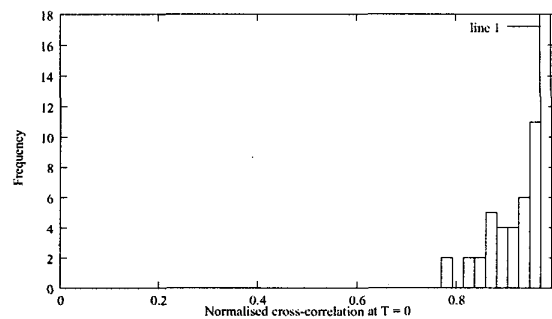
To assess the validity of the existing noise models, simulations with different values of the parameters in Equation 1 have been compared with observed noise signals, such as that shown in Figure 4.

While existing noise models may be approximately valid for noise observed in some types of communications channels, for telephony systems many noise events which are observed cannot be generated with these models. The main reason for this appears to be that the impulses are assumed to be generated at the point of transmission - that is, all impulses seen at the receiver are filtered by the same channel impulse response. This means that the shaped impulses are, in effect, scaled versions of each other. However, it is clear that the filtered impulses which are observed at the receiver are of quite different shapes and lengths.

<sup>1</sup>This data has been kindly provided by Telstra Research Laboratories, Australia



(a) Normalised Reduced Singular Vector



(b) Histogram showing goodness-of-fit

Figure 3: The resulting waveform, which shows an excellent fit for most of the sample waveforms

In reality, only impulses originating at the transmitter are shaped by the response of the entire channel. From studies of local-loop impulse noise data, it is clear that a significant number of impulses have a point of ingress elsewhere on the line, since they have been shaped by different partial channels depending on the precise point of ingress.

It is proposed that in order to generate a manageable model for such a system, the channel should be divided into a finite number of discrete segments. Each of these segments may be considered to have a certain probability of experiencing an impulse during a particular sample period. A simple filter model of the channel segment from this point to the receiver (which may include a number of different types of cable) may then be constructed, based on the distributed RLC model. The synthetic impulse response of each of these filters may be obtained by simulating the impulse response of a truncated RLC ladder using one of the models described in Section 2. This may then be implemented as a digital filter.

Finally, since we must now combine the impulse noise sources with the various channel impulse responses, we can also do the same to the data signal, which will result in a final expression for the signal as given by

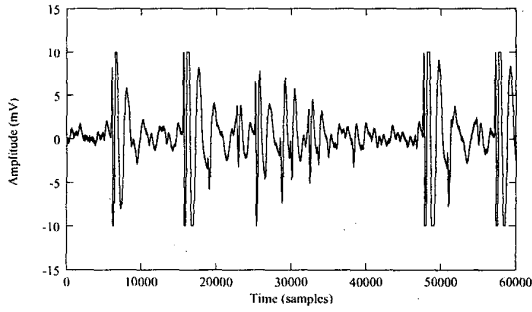


Figure 4: Noise waveform from a real telephone line

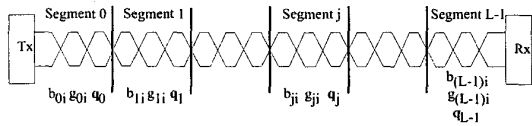


Figure 5: Segmented wire model with parameters attached

$$r(n) = \sum_{k=0}^{N-1} h_k(x(n-k) + G_2 g_w(n)) + \sum_{k=0}^{[n, N-1]} \sum_{j=0}^{L-1} b_{j(n-k)} G_{3k} g_{j(n-k)} q_{jk} \quad (5)$$

In Equation 5,  $r(n)$  is the received signal,  $x(n)$  is the transmitted signal (i.e. the output of the modulator),  $h_k$  is the impulse response of the entire channel,  $G$  is a gain term for the Gaussian white noise, which is represented by the random variable  $g_w$ . The impulse noise model is the most generic case, with  $b_{mn}$  referring to the Bernoulli random variable (i.e. is there or is there not an impulse) for the  $m$ th cable subsection, during the  $n$ th sample.  $g_{mn}$  refers to a gain term, which is a Gaussian or stable-distribution random variable, for the  $m$ th subsection during the  $n$ th sample.  $q_{mn}$  is the  $n$ th filter coefficient for the filter corresponding to the  $m$ th subsection.  $G_{3k}$  is a gain term for impulse noise corresponding to the  $m$ th subsection. These parameters are illustrated in Figure 5.

To assign probabilities for the occurrence of an impulse in a particular subdivision of the cable, it is necessary to look at long continuous recordings of the signals on idle telephone lines. The peak amplitudes of isolated impulses may be tabulated and thus parameters for the Gaussian or Stable-distributed impulse noise terms can be determined. The Gaussian noise component may be determined by looking at a quiet period and estimating the gain and variance.

## 6. Conclusions

A new model for telephone lines which combines the frequency response of the line with the noise model, and which allows for

an arbitrary point of ingress for impulse noise events, has provided a more realistic basis for development and optimisation of ADSL and VDSL systems.

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