During class please **switch off** your mobile, pager or other that may interrupt.

**Course objectives:**
- The trainee will understand the working principles of ADSL.
- The trainee will get an overview of all the problems and solutions when using high bitrates on copper twisted pair.

**Entry level requirements:**
- Intro to ATM (GETE2441) => basic ATM for applications in adsl

**Suggested duration:**
1/2 day  
Normal class hours 08:30h => 12:00h  
13:00h => 16:30h
This presentation must be read in the PowerPoint SLIDE-SHOW mode, because it contains a lot of animation.........
Table of content

- Introduction
- Standards
- Restrictions
- Modulation
- Error detection and correction
- xDSL technologies
Introduction
Frequencies within the voice band are transmitted through the switched connection of a PSTN network.

This voice band is used for voice communication or modem communication (i.e. fax, V.32, V.90, ...)

Modem to modem communication in POTS band through the PSTN network!
DSL technologies use also other frequencies outside the voice band to modulate information on your local telephone line.

- ISDN provides you with a 160 kbps connection on your local line.
- ADSL provides you a high speed connection on your local line.
Today we are facing 2 problems.

(1) The voice network, specifically designed to transport voice (using circuit switching) is not ideal to transport data. Mainly because of the bursty nature of data communication and the limited capacity of the voice network (64 kbps). That is why a B-ISDN was created.

(2) slow access: the capacity of an analogue modem is limited to about 30 kbps.

The solution to this problem is ADSL, where we extend the used frequency band to about 1MHZ.

V.90 technology is able to accelerate data downstream from the Internet to your computer at speeds of up to 56Kbps.
Using ADSL downstream, data speeds up to 8.1 Mbps are possible. The maximum upstream speeds is limited to about 1/10 of the maximum downstream speed.

The distance is however limited to 5.4 km.

ADSL offers the opportunity to run multiple services at the same time: voice, surfing the web, VOD,...
We introduce the concept of FDM (Frequency Division Multiplexing).

Apart from the traditionally used frequencies over the UTP (300 - 3400 Hz), we start using higher frequencies for the ADSL up- and downstream channel.

As ADSL is an asymmetrical service with more capacity in the downstream direction, we need more frequencies in this direction.

It will be explained that the highest frequencies result in problems. The capacity of transporting data decreases with the increase of the used frequencies. In other words it is a pity that the frequencies reserved for POTS cannot be used for ADSL. POTS frequencies cannot be used for ADSL since the Life Line concept is still valid. The latter means that one should be able to make a phone call in case of a power failure.

VoDSL (making phone calls over the ADSL signal) does not support this Life Line concept.

ADSL speeds would increase when ACTIVE filters (splitters) were used. Because of the Life Line concept only PASSIVE are allowed since the active splitters contain OPAMPS that need to be powered.
The lower frequencies used by ADSL can disturb the audible spectrum and need to be filtered out towards the telephone set.

With on-hook and off-hook situations of your telephone the line impedance changes which will have an impact on the ADSL modem communication.

Voice & data are transported over the same copper wire simultaneously in both directions (full duplex). The ADSL signals travel between the Central Office (CO) and the ANT (ADSL Network Termination).

A wide range of filters/splitters is available on the market today adapted to local situations. From country to country complex impedance’s can differ as also the physical presentation of the local line.
Voice/Data over DSL?

- ADSL suitable for all types of communication (voice and data)
- Evolution towards Full Digital Loop (FDL)
  - Elimination of POTS lifeline (AND splitter)

In ADSL the voice is send in a separate part of the spectrum (FDM). The signal remains analog. The disadvantage is the need of a Pots Splitter and the fact that you have only one line.

In the case of VoDSL the telephones (up to 16) are connected to the VoDSL modem and the voice is multiplexed on an ATM connection over ADSL. All 16 phones can be used at the same time for different calls.

VoDSL modems these days implement 4 or 8 telephone ports.

The limitation of 16 is due to the limitation of the upstream ADSL bitrate. In the case of SHDSL the number of phones could be increased up to 32.

In the way the voice and the data traffic are combined. This is voice / data convergence in the access network only.

In most countries local regulations imply that the lifeline service must be available at all times. As our ADSL modem is locally powered a power outage will result in loss of our ADSL connection. This means that we will still need our POTS service even with VoDSL. At the moment R&D centres are looking at remote powering the ADSL modem so that ADSL would keep functioning and VoDSL is still possible. This will then make our POTS band dispensable meaning we would also be able to use those frequencies for xDSL communication.
The slow and inefficient data communication is not the only problem. Since we currently transport all data via the voice network our voice network is becoming overloaded.

(Nice to know that an average phone call lasts about 3 to 4 minutes whereas an average surf session lasts about 20 minutes)

To relieve the voice network we redirect the data in an ASAM (multiplexer) towards an ATM network.

For ADSL modem-to-modem communication we have following concepts:

- **ATU-C & ATU-R**: ADSL Terminal Unit Central and ADSL Terminal Unit Remote
- **ADLT & ADNT**: ADSL Line Termination & ADSL Network Termination

Both concepts apply to the ADSL modem.
ANSI Standards

- **ANSI T1.413 Issue 1** 1995
  - First ADSL specification out in 1995 was STM based and not clearly build

- **ANSI T1.413 Issue 2** 1998
  - Second ADSL specification which was mostly driven by Alcatel and ATM based as is used today.
ITU-T Standards (2)

- **ITU-T G.dmt or G992.1**
  - Specification by ITU-T which is based on the ANSI T1.413 issue 2 standard plus an extra handshaking protocol.
  - Annex A: specifies operation above the POTS band
  - Annex B: specifies operation above the ISDN band
  - Annex C: specifies operation for Japanese ISDN band

- **ITU-T G.lite or G992.2**
  - Specification by ITU-T which is a kind of "stripped down" ANSI T1.413 issue 2 standard plus an extra handshaking protocol.
  - Based on recommendations made by the UAWC workgroup (Microsoft, Compaq & Intel)

- **ITU-T G.hs or G994.1**
  - Specifies the handshaking procedure for DSL transceivers
The split between the frequencies used for the upstream direction and the frequencies for the downstream direction is at 138kHz.
Q: How can we increase the data speed and respect the symbol rate related constraint? (Nyquist)

A: Increase the number of bits per symbol via different modulation techniques like QAM.

Bitrate $\Rightarrow$ expressed in bits per second (bps)
Symbol rate $\Rightarrow$ expressed in baud

How to understand the difference between symbol rate and bit rate?

Assume simple Amplitude modulation $\Rightarrow$ when you want to send digital information over a line you can transmit a bit over the line represented by a certain voltage level, for example $+3V$ to represent a logical 1 and $-3V$ to represent a logical 0.

When representing 1bit by a certain voltage level the symbol rate = the bit rate.

When adding more voltage levels you can actually specify more bits per symbol, for example $+3V$ represents the logical bit-sequence 11, $+1V$ represents logical 10, $-1V$ represents logical 01 and $-3V$ represents the logical bit-sequence 00

Here you have actually put 2 bits in a symbol and that way doubled the bitrate. On the other hand the symbol rate in baud has remained the same.
Nyquist bandwidth constraint

For a given bandwidth (W in Hz), the maximum amount of symbols/second (Rs in baud) is limited in order to avoid Inter Symbol Interference (ISI).

- Each symbol corresponds to a number of bits.
- You need to be able with the current technology to distinguish one symbol from another.
- Also the symbol period is minimum the signal period of the lowest frequency.
The maximum attainable data speed depends on the signal to noise ratio (SNR).

The higher the (allowable) signal strength, and the lower the amount of noise on the line, the higher the capacity.

Unfortunately a lower noise level requires high quality lines which are expensive or not available.

On the other hand the signal strength is limited (governmental constraints) in order to limit the amount of crosstalk.

A decreasing signal to noise ratio will result in more bit errors (BER) on the line but with current technologies it is possible to detect AND correct these errors up to a certain level. It can be said that by introducing these error detection/correction mechanisms we can actually increase the capacity of the line for a certain SNR and BER.
The figure illustrates the distance dependency of the Shannon/Hartley capacity theorem.

Since the attenuation (signal loss) increases with distance (cable length) the maximum data speed drops with the distance.

Theoretically ADSL can reach a downstream capacity of apc. 15Mbps at 0km. Although in practice this is limited to 8,1Mbps.
The signal strength is not only distance dependent. It is also frequency dependent. This is due to the skin effect.

\[ R = \frac{\varphi \times d}{S_{\text{eff}}} \]

- \( R \) = resistance (\( \Omega \))
- \( \varphi \) = resistivity (\( \Omega \)m)
- \( d \) = distance, length of conductor (m)
- \( S_{\text{eff}} \) = effective cross sectional area of conductor (m\(^2\))
What means a loss of 32 dB at 150 kHz?

Attenuation (dB) = 10 x log_{10} (P1/P2)

P1/P2 = Inv log_{10} (Attenuation / 10)

P1/P2 = Inv log_{10} (32 / 10) = 1585

⇒ this means that our received pulse is **1585 times smaller !!**

What means a loss of 55 dB at 150 kHz?

Attenuation (dB) = 10 x log_{10} (P1/P2)

P1/P2 = Inv log_{10} (Attenuation / 10)

P1/P2 = Inv log_{10} (55 / 10) = 316228

⇒ this means that our received pulse is **316228 times smaller !!**
Higher frequencies suffer more from attenuation than lower ones, since the skin effect has more impact on the higher frequencies.

That is why the upstream signal is less attenuated than the downstream channel.
In some countries it is common practice of splicing a branching connection (bridged tap) onto a cable. Thus a bridged tap is a length of wire pair that is connected to a loop at one end and is unterminated at the other end. Approximately 80% of loops in the US have bridged taps; sometimes several bridged taps exist on a loop. One reason for a bridged tap is that it permits all the pairs in a cable to be used or reused to serve any customer along the cable route. Most countries in Europe claim to have no bridged taps, but there have been reports of exceptions.

The reflection of signals from the unterminated bridged taps results in signal loss and distortion.

When 3 is asking for phone call service a main cable is put in the ground.

Later on when 2 and 1 ask the same thing a lateral is drawn from the main cable.

Imagine 2 decides to move out (people in America frequently move) and thus cuts the lateral. Since there is no longer a telephone connected to the lateral there is no nice power dissipation anymore. This results in reflections; reflections that travel in both directions of the main cable.
A telephone cable contains up to several thousand separate wire pairs packed closely together. The electrical signals in a wire pair generates a small electromagnetic field, which surrounds the wire pair and induces an electrical signal into nearby wire pairs. The twisting of the wire pairs reduces this inductive coupling (also known as crosstalk), but some signal leakage remains.

NEXT (near end crosstalk) is a major impairment for systems that share the same frequency band for upstream and downstream transmission. NEXT noise is seen by the receiver located at the same end of the cable as the transmitter that is the noise source. In other words NEXT means that the receiver of one twisted pair picks up noise transmitted (by the transmitter) another one.

Transmission systems can avoid NEXT by using different frequency bands for upstream and downstream transmission. FDM (Frequency Division Multiplexed) systems avoid NEXT from like systems (also known as self NEXT). FDM systems still must cope with NEXT from other type of systems that transmit in the same frequency band, and other phenomenon known as FEXT (far end crosstalk).

FEXT is the noise detected by the receiver located at the far end of the cable from the transmitter that is the noise source. FEXT is less severe than NEXT because the FEXT noise is attenuated by traversing the full length of the cable.

A major advantage of fibre optic transmission is the lack of any crosstalk.

Conclusion: NEXT is worse than FEXT for systems sharing the same frequency band in the up and downstream direction. Alcatel ADSL uses FDM and thus avoids NEXT. Here FEXT becomes the critical impairment.

When ADSL is mixed with other systems in the same cable then NEXT can occur because of overlapping frequency ranges! (see next slide)
When AoP (ADSL over POTS) and AoI (ADSL) over ISDN reside in the same cable there is NEXT.

Some frequencies of the downstream transmitter of an AoP line overlap with the receiver frequencies of an AoI line.

ADSL over an ISDN line uses another spectrum than ADSL over a POTS line. When both systems are used within the same bundle of a cable Near End Crosstalk (NEXT) will occur.
Modulation
Looking at an analogue signal, described by means of a sine function, modulation techniques exist by varying amplitude, phase, frequency or by a combination of these.

QAM is a modulation technique where both amplitude and phase are modified.

The amount of bits we can put on 1 symbol depends on the amount of amplitude and phase levels we distinguish. The latter is reflected in the constellation grid as shown on the slide.

Since 16 points are distinguished there are 16 different combinations of amplitude and phase.

The amplitude is the length of the vector whereas the phase is measured counter clockwise from the $x$-axis towards the vector.

In the example we can put up to 4 bits in 1 symbol, or in other words 4 bits are needed to construct 1 symbol. (4 bits allow you a specific bitsequence for each of the points $(2^4)$in the constellation)

Increasing the number of bits per symbol results in a higher attainable data speed.
Because the transmission lines are far from perfect noise is picked up during transmission. This results in a deformation of the analogue signal. In other words the analogue signal arriving at the other side of the transmission line might have a (slightly) different amplitude and phase.

From the constellation grid one can see that if the deformation of amplitude and/or phase is too big, mistakes might occur.

There is always 1 dot in the constellation grid that is closest to the reconstructed vector at the receiving end.

Imagine that 1001 was intended but that because of deformation the reconstructed vector is closer to 1011, the latter one is thought of to be the correct one.

Expanding the constellation grid would fulfil 2 desires:

1) Increase the data speed by putting more dots in the constellation grid. But because the grid will become more dense it is much more likely to make mistakes. Expanding the grid is a solution.

2) Imagine that I would like to put the same amount of bits per symbol under all line conditions. The more noise, the more mistakes! Expanding the grid, depending on the line conditions would be ideal!

Unfortunately the signal strength (signal power,amplitude) is limited because of cross talk limitations. This limitation corresponds with a maximum radius of the circle that describes the amplitude of vectors constructed in the constellation grid.

That is why noise and attenuation are measured first to determine how many bits one can put on the line.

If we take a closer look at the dots in the constellation grid, we see that 2 adjacent dots do not differ more then 1 bit from each other. Just to make sure that in case of an error the error is small (single bit error). Dots will differ in more bits when further apart in the constellation.
The table can be used in 2 ways:
(a) what is minimum required SNR to modulate N bits on a carrier
(b) how many bits can be modulated given a SNR of Y dB

<table>
<thead>
<tr>
<th>Bits/symbol</th>
<th>QAM</th>
<th>Signal/Noise ratio (dB) for BER&lt;10^{-7}</th>
</tr>
</thead>
<tbody>
<tr>
<td>4</td>
<td>QAM-16</td>
<td>21.8</td>
</tr>
<tr>
<td>6</td>
<td>QAM-64</td>
<td>27.8</td>
</tr>
<tr>
<td>8</td>
<td>QAM-256</td>
<td>33.8</td>
</tr>
<tr>
<td>9</td>
<td>QAM-512</td>
<td>36.8</td>
</tr>
<tr>
<td>10</td>
<td>QAM-1.024</td>
<td>39.9</td>
</tr>
<tr>
<td>12</td>
<td>QAM-4.096</td>
<td>45.9</td>
</tr>
<tr>
<td>14</td>
<td>QAM-16.384</td>
<td>51.9</td>
</tr>
</tbody>
</table>
Discrete Multi Tone (DMT)

- For ADSL multiple carrier frequencies are modulated on the 1 ADSL line using Quadrature Amplitude Modulation.
- These frequencies are equally spaced and for each carrier the SNR is measured to determine the maximum achievable QAM.
- The sum of all frequencies is put on the line.
- This principle is called Discrete Multi Tone (DMT).
Imagine a certain frequency spectrum that is divided in 3 subchannels. To every of these 3 subchannels we can assign an appropriate QAM scheme depending on the measured SNR. The sum of these QAM signals is made and then sent to an DAC (Digital Analogue Convertor). The outcome is an analogue signal that is put on the transmission line.

Dividing the frequency spectrum in subchannels can be done in different ways.

Or we can consider only a few subchannels but then we loose the flexibility of assigning different amount of data bits to relatively small fractions of the frequency spectrum; or we could consider an enormous amount of subchannels. This results in increased complexity.

255 subchannels is a nice trade off.
DMT and ADSL

- The spectrum used for ADSL is divided into 255 carriers.
  - Each carrier is situated at \( n \times 4.3125 \text{ kHz} \)
- For the upstream direction carriers 7 to 29 are used
- For the downstream direction carriers 38 to 255 are used
- On each carrier the SNR is measured and the QAM determined.
  - Minimum \( \Rightarrow \text{QAM-4} \Rightarrow 2\text{bits/symbol} \)
  - Maximum \( \Rightarrow \text{QAM-16384} \Rightarrow 14\text{ bits/symbol} \)
- Symbol period for each carrier: \( \pm 250 \mu\text{s} \)
Since the attenuation increases with the frequency (skin effect) the SNR drops with increasing frequencies.

Thus less bits/carrier can be assigned for the upper subchannels.

This explains why it is useless to consider frequencies above 1.1 MHz.

Considering frequencies above 1.1 MHz is done in VHDSL (Very High Speed DSL). Since we have to compensate for the skin effect, in this application the distance has to be limited.

What about Discrete Multitone (DMT)?

Dividing the used frequency spectrum in subchannels results in the possibility to assign a different QAM scheme on every subchannel. So depending on the SNR of a specific carrier, more or less data bits can be transported.
We will always assign less bits to the carriers than allowed by the measured SNR. Typically we assign an average of 2 bits less.

This margin is called the Target Noise Margin and is configurable via the AWS. You specify what the average margin should be right after startup in dB. The modem measures the available SNR, then subtracts the Target Noise Margin, and then checks to what constellation it would fit. By default the TNM is 6 dB.

Why?

In case of a disturbance (e.g. RFI), we do not want our overall speed to drop. Whenever a subchannel becomes unavailable to transport data bits the spare bits in adjacent subchannels will be used. (see next slides)
After startup we will use a lower QAM then possible on most of the carriers.

- The measured SNR at startup determines the maximum possible QAM at startup, e.g.: QAM-4096 corresponding to 12bits/symbol
  >>> used QAM on that carrier: QAM-1024 (10bits/symbol)
- This results in extra bits that could be allocated on that carrier.

During showtime (modem operation) the SNR is measured on all carriers at regular intervals (default 1sec).

- If the SNR on a certain carrier degrades resulting at a lower QAM that can be used on that carrier, the bits of that carrier will be re-allocated to other carriers where the max QAM is higher then the actual used QAM.
- The modems will try to spread out the re-allocated bits over numerous carriers.

It is not fully correct to speak about QAM constellations and bits here. It should be noise margins. The rule is: the ATU tries to equalise the noise margin over all tones, what could lead to bits being moved from one tone to another, and even tones to disappear or to appear. Margins are "analog" where bits are "digital". As you need 3 dB for 1 bit, a margin of 4.7 dB over all tones can not be translated into bits.

Modem does not calculate with bits. It measures the available SNR, then subtracts the Target Noise Margin, and then checks to what constellation it would fit (table slide 31). There is also the possibility to increase (or decrease) the gain of each tone (+/-2.5 dB, but sum over all tones must be 0) for those tones which have just a little bit shortage to have 1 bit more loaded. A gain change results directly in signal change, so also SNR change.
Bits swapping explained

Sudden frequency interference decreases SNR on a number of carriers

Bits/carrier

Current max. bits/carrier

Current used bits/carrier

Carriers

Alcatel University Antwerp, page n° 38

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Bitswapping explained

A lower SNR also lowers our max QAM (the number of bits on those carriers)

Current max. bits/carrier
Current used bits/carrier

Bits/carrier

Carriers

Affected frequencies
Bitswapping explained

Bits/carrier

Carriers

Current max. bits/carrier

Current used bits/carrier

Current max. bits/carrier

Current used bits/carrier
The ATU tries to equalise the noise margin over all tones.
Communicating modems

ADLT DMT FRAME
(ATU-C)

DOWN modulation

UP demodulation

ADNT DMT FRAME
(ATU-R)

DOWN demodulation

UP modulation

Unshielded Twisted Pair

2 analogue signals (UP, DOWN) travelling in opposite directions over the UTP

♦ ATU-R locks to the pilot carrier in the downstream direction (PLL)

➢ The better the lock the better the overall SNR!

Modems using the ANSI ADSL standard locks to a carrier in the up- and downstream direction, i.e. the ATU-R locks to the pilot (carrier 63) in the downstream part and the ATU-C locks to the pilot in the upstream part. ANSI has 2 pilot carriers, 1 in the downstream and 1 in the upstream.

Modems using the ITU-T ADSL standards only require the ATU-R to lock to the ATU-C and also use this lock for his upstream transmission.
ADSL Superframe

- **DMT Symbol:**
  - A DMT symbol is the sum of all symbols on each individual carrier.

- **Data Symbol (DS):**
  - A data symbol is used to transmit information (data).

- **Synchronisation Symbol (SS):**
  - A sync symbol is transmitted after 68 data symbols to assure synchronisation and to detect a possible loss of frame.

- **ADSL symbol period:**
  - $T_s = \frac{17 \text{ ms}}{69} = 246.377 \mu s$
  - $T_s = \frac{17 \text{ ms}}{68} = 250 \mu s$ (symbol period for the data plane!)
Error detection and correction
**Error detection & correction principle**

Imagine that the green spots on the slide represent a bitsequence that is used to transmit data. If during the transport bit errors occur (single or multiple) they will result in one of the red invalid bitsequences. At that moment the receiver can execute bit error correction by locating the nearest green spot (see example above). In this way it is possible to correct up to 2 bit errors.

If a 3 bit error occurs, it cannot be corrected because then the distance to the green spots (left and right) is equal. In this case only an error detection is possible. Even worse, when more than 3 bit errors occur, it is possible that the other side receives a “red” bitsequence that is closer to another green spot. In this case a wrong correction is executed and the error correction code fails.

The error correction possibility can be calculated as follows:

- DISTANCE between the green spots = $6 = 2t + 1$
- Number of errors than can be corrected = $t = 2$

The distance between the valid data sequences can be increased by adding check bytes to the original data (see Reed Solomon, Trellis, …)

**Error detection principle**

In the previous example a 3 bit error was the maximum allowed. More than 3 bit errors will result in wrong corrections. It is possible to increase the number of bit errors that can be detected in a simple way. If both sides agree not to execute error corrections, up to 5 bit errors can be detected (because the distance is 6). In other words, all red spots will result in an error detection. The disadvantage is that there is no error correction possible.

**Examples**

- ADSL (Reed Solomon and Trellis) always works in correction mode
- ATM (HEC code) uses correction mode and will temporary switch to detection mode when errors occur.
Error detection & correction

♦ The choice of correction mode or detection mode is a consensus!

➢ Will you choose the ability to detect AND correct errors with the possibility you introduce more errors,
➢ Or will you choose the ability to detect a higher number of errors with no possibility to correct them but at least you not introduce more errors.

♦ Some coding mechanisms will switch from correction mode to detection mode as soon as errors are detected! (e.g. ATM)

➢ This is done done as errors are mostly of a bursty nature and never come alone!
Because of the presence of impulsive noise (infrequent high amplitude bursts of noise mostly caused by central office switching transients, dial pulses, POTS ringing, neighboring railway stations, elevators, ... that can corrupt the signal beyond recognition) on the twisted pair wire means must be implemented to make sure that the ADSL transceiver is sufficiently robust against this impulsive noise and to maintain an acceptable BER for good quality of services.
Since errors often come in bursts it is likely that the error correction possibility is unsufficient.
Interleaving:

In stead of transmitting our RS code words directly on the line we will create a frame of the same size made up by multiple RS words by taking only a portion of each of the original RS words. This has the advantage when a burst of errors occur on the line and the original RS words are recreated on the receiving side, the errors will be spread over multiple RS words. This could mean that we are able to correct the errors within a single RS word if the number of errors are within the RS correction boundaries.

Interleaving depths:

The main disadvantage of interleaving is the high delay. Constructing the blocs that will finally be transmitted over the line take time as you have to wait for a time before you can actually start transmitting. In our example here we need 15 original RS words (1 Byte of each) before we can construct the first block that is actually transmitted over the line.

At the receiving side it will also cost extra time to reconstruct the original RS word. The first original RS word can not be reconstructed before we have received all the bytes of this first RS word.

Interleaving can be sped up by using different depths, i.e. by taking bigger chunks of the original RS words you will be able to construct your first bloc for transmission much quicker. This has the disadvantage that errors will be spread over less RS words on the receiving side with the possibility that they can not be corrected.

Alcatel introduces 3 interleaving depths: high, medium and low.

Where high uses the smallest portions to construct a frame for transmission, consequently the high interleaving depth will introduce the highest delay as this one will need more frames to construct/reconstruct before transmission/reception.
RS has the characteristic of generating bursty ATM traffic
  - The result might be that a policing function discards ATM cells
  - Therefore we need to shape the ATM traffic
    - ATM cells will be transmitted at line rate
Delay & interleaving depths

\[ \text{FAST} = \text{NO INTERLEAVING}! \]
Trellis coding is another error detection and correction mechanism which is optional for ADSL.

Trellis principle =>
- looking at the complete data you're able to detect and correct errors, analogue to detection and correction in spoken language
- example:
  
  | transmitted data: | the water is wet and cold |
  | received data:    | the water is let and cold |

- By just looking at the word *let* we can not know that it is wrong!
- By looking at the information before and after the word we can safely say that it should be *wet* instead of *let*.

Trellis will also add extra information (overhead) to the transmitted data in order to perform his detection and correction.

This overhead is apc. 1/2 bit per used carrier + 4 bits.

A Viterbi coder/decoder will look at the received data and calculate what the most logical value should be for the received data similar to spoken language as illustrated on the slide.
ADSL & Reed Solomon

With Trellis used; extra bytes will be needed in the DMT symbol.

For fast (no interleaving) there is exactly one RS word in one DMT symbol. Reed Solomon always performs his calculation on a word of 255 Bytes which is also the maximum length. Our user data (ATM cells) always run through this Reed Solomon coder giving you the maximum speeds as calculated on the slide.

Some payload Bytes of the RS word are used for maintenance communication. Over these Bytes, which are negotiated at start-up, following information can be transmitted

1) EOC: Embedded Operations channel
   Is intended to exchange physical layer messages between the ATU–C and the ATU–R.
   For example what the SNRs are, or what the output power is of the transmitter, ...

2) AOC: ADSL Overhead Control channel
   The ADSL Overhead Channel is intended for bit swap operations

3) CRC: Cyclic Redundancy Check
   The transmitter shall calculate a CRC over the data of last superframe, and send this CRC to the receiver. The receiver also calculates the CRC over the same data block, and compare it to the received CRC value. When those do not correspond, a CRC error event shall be generated.

4) IB: Indicator Bits
   They indicate a number of states and alarms like CRC errors, loss of signal, loss of frame, etc.

5) Dummy information
From the table on slide 31 we have seen that to attain a BER of $10^{-7}$ for a specific QAM you need a certain SNR.

- If the SNR is lower then this value the BER becomes too high.
- By introducing error detection and correction you lower the BER because you correct a number of the introduced errors.

The mechanism introduces a coding gain resulting in an actual lower SNR that is needed to achieve a certain constellation.

- Trellis introduces a coding gain of apc. 5.5dB.
- Reed Solomon introduces a coding gain of apc. 4dB.
- Trellis & RS together introduce a coding gain of apc. 9dB.

The numbers given a relative and not exact. In ADSL implementations the numbers will differ a bit depending on the QAM desired and the product manufacturer.
Relative Capacity Occupation (RCO in%): proportion of the used line rate to the attainable line rate

Reed Solomon has to be supported by the ADSL modems according to the standardisation. This does not mean that Reed Solomon must be activated at all times.

Trellis on the other hand is optional!

During start-up of the ADSL modems capabilities are transferred between the modems and based on this the ATU-C (in the ASAM) decides what is used. Also during this phase it is communicated how many bytes are used in the framing overhead, this is mostly only 1 Byte to have the most optimal performance.

Example for downstream: (taken from ADSL diagnostics)

<table>
<thead>
<tr>
<th>ATM attainable rate:</th>
<th>8,128 Mbps [((255-1) x 4000) x 8]</th>
</tr>
</thead>
<tbody>
<tr>
<td>ATM used rate:</td>
<td>8,128 Mbps</td>
</tr>
<tr>
<td>Used line rate:</td>
<td>8,612 Mbps</td>
</tr>
<tr>
<td>Attainable line rate:</td>
<td>10,236 Mbps</td>
</tr>
<tr>
<td>RCO:</td>
<td>84% [((8,612 / 10,236) x 100]</td>
</tr>
</tbody>
</table>

Short haul local loop with Reed Solomon disabled (used ATM rate = attainable ATM rate) and Trellis activated (Used line rate >> {ATM used rate + modem-modem communication overhead}). And there’s 1 Byte of framing overhead.
xDSL Technologies
Over the years many different symmetrical DSL types have been defined by different standardisation bodies. Where to place these different types, with which speed, over the years you can see on the slide. Most of them are designed for a set fixed speed indicated by the small squares. Only SDSL and SHDSL are rate adaptive, meaning that the supported speed varies from a minimum to a maximum speed. Depending on the line quality a certain speed can be attained.
Comparison of a few symmetrical DSL technologies. The differences are mainly in the used type of modulation and if it is rate-adaptive or not.

With power backoff we mean the possibility to lower the nominal output power of the transmitter and as such possibly lower the crosstalk.
SHDSL is defined by the ITU-T as Single-pair High speed DSL and is designed primarily for duplex operation over two-wire twisted metallic pairs. Optional four-wire operation is supported for extended reach applications. SHDSL transceivers are capable of supporting selected symmetrical user data rates in the range of 192 kbps to 2.312 kbps using a Trellis Coded Pulse Amplitude Modulation (TC-PAM) line code. They are designed to be spectrally compatible with other transmission technologies deployed in the access network, including other DSL technologies. SHDSL transceivers do not support the use of analogue splitting technology for coexistence with either POTS or ISDN.

Symmetrical and assymetrical PSD masks are defined but only the symmetrical PSD masks are mandatory. The assymetrical PSD masks are optional and are for the North American market. Currently 2 assymetrical PSD masks have been defined for following datarates:

1) 768 kbps or 776 kbps
2) 1536 kbps or 1544 kbps.

The exact meaning of PSD is explained further in the presentation!
Any SHDSL modem has these main functions. The user data is mapped into the SHDSL frame (explained on next slide) after which the frame is modulated using TC-PAM 4. TC-PAM 4 means that in each symbol we can put 4 bits, meaning there are 16 different distinctive levels. Of the SHDSL frame we take 3 bits to put into the symbol the fourth bit in the symbol is used by the trellis coder to put his coding bit (Trellis overhead).

We can say that the Trellis encoder adds a coding bit every 3 bits in the SHDSL frame and then we take these 4 bits to create the corresponding symbol and put it on the line.
The payload rate is requested by the user and is determined by the formula mentioned on the slide. When \( n = 36 \) the value for \( i \) is limited to 0 or 1.

With current implementations for transport of ATM cells (like in ADSL) the value of \( i \) is limited to 0!

The transmission of the last 4 stuffing/spare bits is determined by the transmission mode, either we transmit 0, 2 or 4 of these bits. In case of 2 bits, with synchronous transmission mode (explained on next slide) these are spare bits and the exact length of a SHDSL frame is 6ms.

Within a SHDSL frame we find a frame sync word used for frame synchronisation, overhead bits (usage explained on next slide), 4 payload blocks of a size \( K \) and up to 4 extra bits depending on the transmission mode.

The size of each payload block is also determined by the parameters \( n \) and \( i \), and so in relation with the requested user data rate.

The higher the requested user rate the longer a SHDSL frame will be. We can clearly see with this that for a higher user rate we need a higher symbol rate because of the longer SHDSL frame.
Plesiochronous mode uses a clocking scheme in which the SHDSL frame is based on the input transmit clock but the symbol clock is based on another independent clock source.

Synchronous mode uses a clocking scheme in which the SHDSL frame and the symbol clock are based on the STU-C input transmit clock or a related network timing source.

Synchronous mode is mainly used by the current implementations.
The overhead bits (OH) in the SHDSL frame transport:

- **Embedded Operations Channel (EOC): 20 bits**
  - Transports maintenance messages over HDLC protocol

- **Cyclic Redundancy Check (CRC-6): 6 bits**
  - CRC over transmitted SHDSL frame

- **Indicator Bits (IB): 4 bits**
  - To report loss of signal, segment anomaly, power loss & loss of sync

- **Stuffing indicator bits: 2 bits**
  - Only used in plesiochronous mode (spare in synchronous mode)
  - Indicate presence of stuffing bits at the end of SHDSL frame

The 32 overhead bits within the SHDSL frame are used for following purposes:
SHDSL rates calculation

- SHDSL frame length = 14 + 32 + 2 + (4 x K) bits
  \[= \text{sync word} + \text{OH} + \text{spare} + (4 \times \text{Payload blocks } K)\]
- K = 12 (8n + i) bits
- Datarate R = 64n + 8i kbps

Frame is Tx in 6 ms, from this we calculate the symbol period (Ts)

- With 3 bits per symbol each symbol takes
  \[Ts = \frac{6\text{ms}}{\text{frame length}/3} = \frac{6\text{ms} \times 3}{48 + (4 \times 12 \times (8n+i))}\]
- \# symbols / second = 1/symbol period = Rs = 1/Ts
  \[= \frac{48 + (4 \times 12 \times (8n+i))}{6 \times 3}\]
  \[= \frac{8 + (8 \times (8n+i))}{3} = \frac{8 + 64n + 8i}{3}\]
  \[= \frac{8 + R}{3} \text{ ksymbols/sec}\]

- Rs = \[8 + (R \text{ in kbps})\] / 3 in kilosymbols / second

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As said on the slide the power spectral density (PSD) is a good representation of the associated energy that a specific type of modulation has within a certain frequency band.

This PSD gives us a good indication for what frequencies there’s a stronger radiation. This is especially important for field implementations where we have cable bundles and where we should limit the radiation from one cable to the other and as such the crosstalk.

SHDSL is a rate adaptive technology using TC-PAM which means that for higher data rates the used frequency band is also higher. (symbol rate is higher!) We can see on the slide that for a data rate of 768kbps the PSD affects mostly the frequencies up to 300kHz and thus interfering with the upstream band of ADSL. In order to limit the crosstalk the different standard organisations have specified PSD masks for the different possible datarates. For a specific data rate the PSD has to stay within his mask. Up to now two types have been defined: symmetrical PSD masks and asymmetrical PSD masks which is mainly used for North American implementations and only for a few distinctive set rates.

The symmetrical PSD masks have to be supported by all SHDSL modems, where the asymmetrical PSD mask are optional.

This drawing represents the symmetrical PSD masks defined by the ITU-T for the datarates mentioned.
VDSL

♦ Very high bitrate Digital Subscriber Line

♦ Assymetric service providing:
  ➢ Upto apc. 52Mbps in the downstream direction
  ➢ Upto apc. 16Mbps in the upstream direction

♦ Uses frequencies upto 30MHz

♦ Distance limited to a few 100 meters

♦ Mainly developed to support 2 High Definition Television channels
Which DSL to choose

Residential customers
• Single phone line
• Internet access

Business customers
• Multiple phone lines
• Network access

Video Services & MTU