An Introduction to G.998.4: Improved Impulse Noise Protection

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A daunting challenge that confronts DSL service providers worldwide is the issue of providing residential and business customers with triple-play services over xDSL that assure IPTV Quality of Experience (QoE). IPTV QoE is defined as the measure of end-to-end performance at the services level from the user's perspective and is an indication of customer satisfaction for an operator's network.

One technique that allows DSL service providers worldwide to deliver advanced triple-play services with assured IPTV QoE is to use **Ikanos' Retransmission Technology**, which dramatically enhances reliability. With this advanced technology, telcos now have an incentive to accelerate the deployment of bundled premium services.

Ikanos' Retransmission Technology is of paramount importance for telephone companies that want to enhance services and efficiently utilize their local loops. This technology is responsible for providing robust designs that maintain optimum throughput with no (or significantly reduced) link drop, downtime, link degradation or retrains.

This whitepaper examines the implementation of Standard **Retransmission System (G.998.4)** as a part of the innovative **Ikanos Quality Video (iQV™) Technology**, which is used to ensure the correct reception of data that is affected by impulse noise. **Ikanos iQV™** is an integrated, automated and intelligent link-reliability solution that assures exceptional quality of triple-play applications that are affected by the presence of various stationary and non-stationary impairments, including impulsive noise.

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1 Introduction

Digital Subscriber Line (DSL) technologies have to contend with a variety of noise sources such as thermal noise, crosstalk noise, radio frequency interference (RFI) and impulse noise. The most widely deployed DSL systems today are compliant to ADSL2 (ITU Recommendation G.992.3), ADSL2+ (ITU Recommendation G.992.5) and VDSL2 (ITU Recommendation G.993.2), all of which recommend discrete multitone (DMT) modulation.

DMT modulation provides modems with significant flexibility to optimize the throughput under a variety of noise conditions. Using DMT, the modulation (or capacity) of each sub-carrier can be tailored individually to the channel and noise conditions that exist in a narrow frequency range of the in-band spectrum. This provides a natural way to address frequency-dependent (but stationary) noises such as crosstalk and RFI.

However, impulse noise is not a stationary noise and the modems require a dedicated approach to mitigate the effects of impulse noise. From the beginning, both ADSL and VDSL technologies have included Forward Error Correction (FEC) mechanisms to provide robustness in a noise environment with impulsive noise. An alternative approach to addressing the effects of impulse noise in DSL systems is to use **Retransmission Techniques**, which offers some advantages and tradeoffs relative to the FEC approaches. Retransmission is the core element in ITU-T Recommendation, G.998.4. It is this standard that defines retransmission as an alternate tool to enhance protection against impulse noise.

In this whitepaper, we address the following topics:

- An overview of impulse noise and existing FEC solutions that are employed in today's DSL modems
- General concept of retransmission
- An introduction to the new ITU-T Recommendation G.998.4 (formerly G.inp) that defines an improved method for Impulse Noise Protection (INP) based on the use of retransmission.
- Highlights of the forthcoming G.998.4 recommendation

2 Impulse Noise

2.1 Causes of Impulse Noise

Impulse noise is a bursty noise event, typically of electromagnetic origin, with potentially high amplitude (power) but limited in duration. The impulse noise is characterized statistically through its amplitudes, duration, inter-arrival times, and frequency spectrum. The causes of impulse noise in the home are multifold and can be introduced in the loop either by man-made and/or natural electromagnetic events:

- Electromagnetic couplings of signals from twisted pair wires (used for home power line wiring) associated with the DSL system
- Electrical equipment used at home, such as hair dryers, vacuum cleaners, light switches and dimmers
- Changes in atmospheric conditions, such as lightning strikes, that affect the signal in the access cable

2.2 Impulse Noise Categories

Impulse noise events exhibit a wide spectrum of durations, amplitudes and inter-arrival times. Although several attempts have been made to characterize the various distributions, impulse noise remains, in essence, a random event that can not be absorbed in the modems' steady-state bit loading. However, for engineering and standardization purposes, two broad categories of impulse noise have been introduced, commonly referred to by the cleverly devised acronyms of "**REIN**" (Repetitive Electrical Impulse Noise) and "**SHINE**" (Single High-level Impulse Noise Event).

- **REIN** events occur with a predictable frequency, which is twice the frequency of the current of the AC-mains. In North-America for example, REIN events are anticipated in a frequency of 120 Hz (the AC current being 60 Hz). In Europe and other regions, the REIN events are anticipated in a frequency of 100 Hz (the AC current being 50 Hz).

  This correlation to the AC cycle indicates that REIN events are, in general, caused by electrical equipment connected to the power mains. The impulses on the power line medium couple to the DSL medium through radiation or other coupling mechanisms. With this affiliation, REIN events are expected to be prevalent in the home environment. The amplitude and duration of REIN impulses are difficult to characterize and show great variation.
- **SHINE** is a catch-all category for isolated impulses that are not predictable in their inter-arrival times. The SHINEs include a wide variety of amplitudes and lengths.

2.3 Impact of Impulse Noise on DMT Symbols

Impulse noise causes significant impairment for any type of communication system, but it particularly impacts DMT modulation.

DMT modulation partitions the incoming data bit stream into blocks of bits. The number of bits in a block (or "data frame") is determined by the prevailing channel and noise conditions during initialization and is updated during Showtime, as needed. Each data frame is modulated on a DMT symbol in the frequency domain and converted to the time domain for transmission over the channel.

Because data is transmitted in blocks, the impact of an impulse noise event is not limited to the instant and to the duration of the actual impulse. A short impulse noise event will typically corrupt an entire DMT symbol and possibly an adjacent symbol, even if the impulse is shorter than the DMT symbol itself. In practice, one assumes that any impulse noise event will corrupt an integer number of DMT symbols. This phenomenon is known as “spreading”.

Protection against impulse noise is clearly imperative. However, requirements have become even more stringent with the evolution of new services offered over DSL. Services that are limited to data transfer have less stringent QoS requirements than those for video services, which is why DSL systems that deliver video services mandate greater protection against impulse noise than data-related services. Various impulse noise mitigation techniques that can be used in DSL systems are described in the subsequent sections.

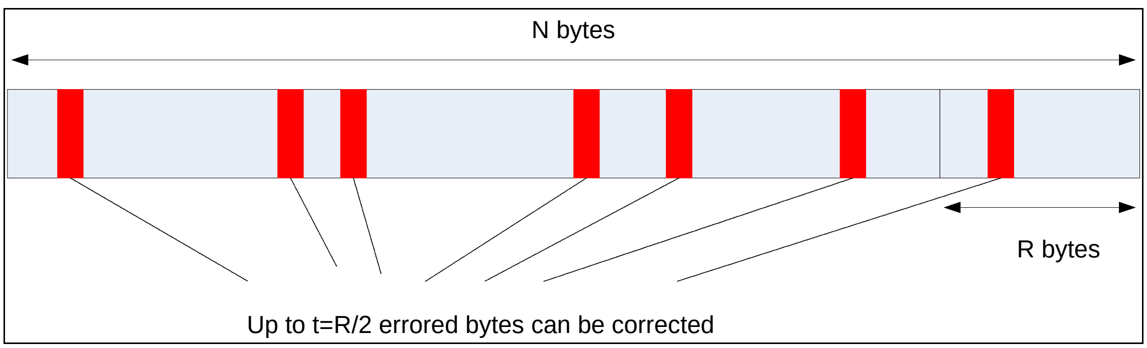
3 Impulse Noise Mitigation Techniques

3.1 Legacy Impulse Noise Mitigation Scheme: FEC and Interleaving

Both ADSL and VDSL support a concatenated coding scheme that consists of Trellis code modulation-as the inner code, and Reed-Solomon coding (RS)-as the outer code. In the absence of impulse noise, these codes are used to provide coding gain to allow for higher bit loading. The coding scheme is also referred to as Forward Error Correction (FEC) since correction of possible errors does not require any additional communication between transmitter and receiver.

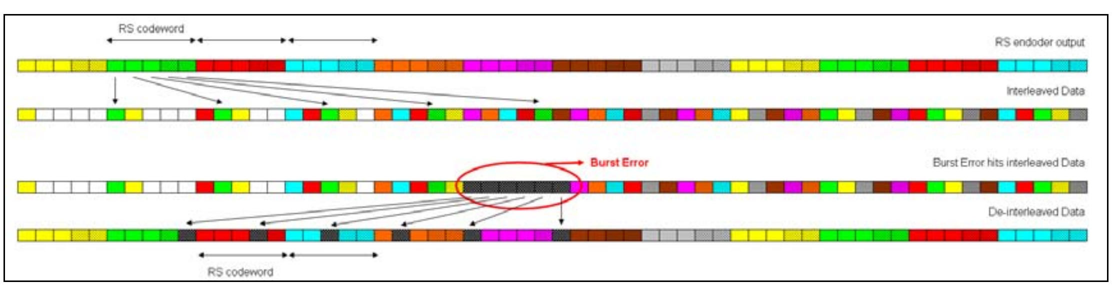
Before the introduction of retransmission, the Reed-Solomon outer code was also the main technique used to protect against impulse noise in ADSL and VDSL. Reed-Solomon codes are block codes that consist of codewords that are N bytes long, of which R bytes are redundancy bytes. **Figure 1** illustrates an RS codeword where, for a given choice of N and R, the Reed-Solomon codes can correct up to t=R/2 bytes in error per codeword[1].

Figure 1: Reed-Solomon Codeword



An impulse will typically completely corrupt many codewords inside an affected DMT symbol. However, the correction capability of the RS code offers inadequate protection against the damage from impulse noise. Coding by itself therefore, does not provide a significant benefit in terms of impulse noise protection. Effective correction and protection against impulse noise may be obtained by combining the RS code with an interleaver at the transmitter and a corresponding de-interleaver at the receiver. This is illustrated in **Figure 2**.

Figure 2: Impulse Noise Protection used on Interleaved and Block Code



[1] The number of correctable bytes can be increased up to t=R when erasure decoding is used, that is, when the RS decoder has information regarding the location of corrupted RS codewords.

Figure 2 illustrates the interleaving and de-interleaving process:

1. After RS encoding, the bytes are interleaved at the transmitter and transmitted in this interleaved order. In other words, bytes are interleaved before transmission. Codewords are represented as groups with the same color.
2. An impulse that impacts the line during this time will corrupt a number of consecutive bytes of the interleaved stream. The number of corrupted bytes is typically too large to be corrected if all bytes would belong to a single RS codeword.
3. However, the bytes still must pass the de-interleaver before the decoding process. During the de-interleaving process, the corrupted bytes are 'spread out' over a number of different RS codewords and 'mixed' with bytes that were correctly received. It is this Spreading Process that allows the number of corrupted bytes, in any given RS codeword, to be within the correction capability of the RS decoding process.
4. The interleaving and de-interleaving result in a single corrupt byte per RS codeword. Assuming that the correction capability (t) is at least one byte, the byte stream will appear error-free at the output of the RS decoder.

3.1.1 Pros and Cons of the FEC and Interleaving Technique

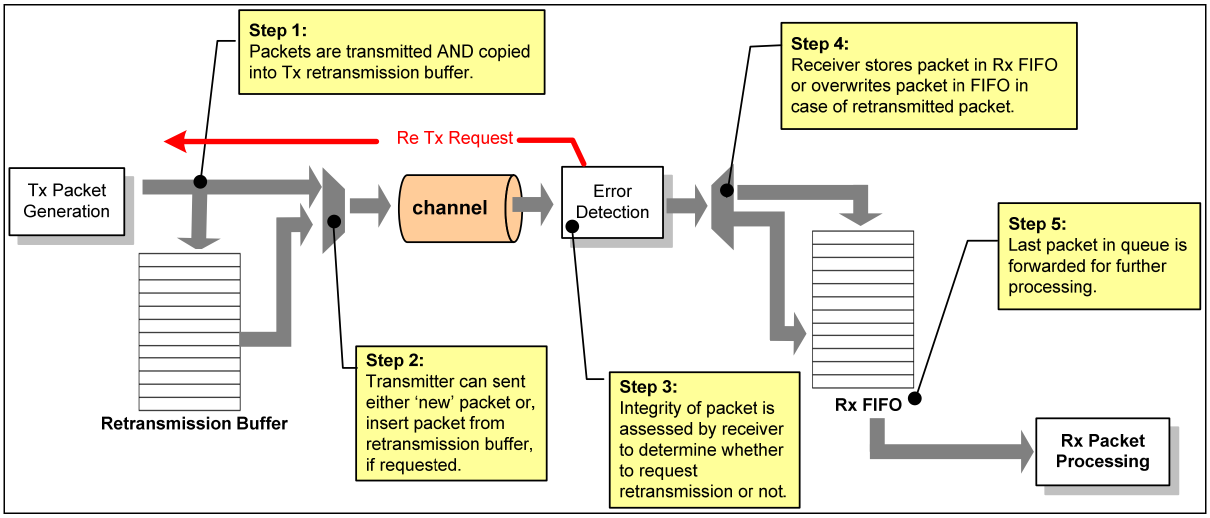
While the FEC and interleaving technique provides an effective mechanism for impulse noise protection, it does present a number of drawbacks. The interleaving that is required to 'spread' the bytes introduces additional delay into the data stream that may not be acceptable for delay-sensitive services. Also, the protection against impulse noise requires a fixed redundancy per codeword. This redundancy is introduced regardless of the occurrence of impulse noise. The longer the maximum impulse length for which protection is required, the higher will be the resultant coding overhead. For long impulses under strict maximum delay requirements, the overhead could climb up to 50%, even if the worst-case impulse occurs infrequently. Impulse noise protection using FEC and interleaving is best suited for frequently occurring impulses of short to medium duration. REIN would be a good example of such an impulse noise environment.

3.2 Retransmission: General Principles

FEC and interleaving rely on the corrective capabilities that are inherent in the DSL coding scheme. Retransmission is a well-known alternative for error correction that has been in use for a long time in a number of other communication systems. In the case of retransmission, data is transmitted in blocks that are structured to allow the receiver to detect correct reception of a block.. When the receiver detects a corrupted block, it communicates this corruption to the transmitter. The transmitter can then re-transmit the same block. The receiver is responsible for buffering any out-of-sequence blocks until a retransmitted block is received. Similarly, the transmitter is responsible for storing any blocks that have not been acknowledged by the receiver; blocks which may consequentially be eligible for retransmission requests. The protocol for acknowledging the reception of data blocks may be a positive or negative acknowledgement based or a combination of the two.

Figure 3 illustrates the basic steps in retransmission.

Figure 3: Basic Steps in Retransmission



In **Figure 3**, the following steps are illustrated:

Step 1: The transmitter partitions the incoming data into structured packets. The packets contain packet identifying information, for example, a sequence number, and some form of redundancy check bytes to allow for error detection. The transmitted packets are copied into a retransmission buffer at the transmitter

Step 2: Instead of assembling a new packet from the incoming data stream, the transmitter may, instead extract a packet from the retransmission buffer..

Step 3: At the receiver, each packet is verified for integrity. If an error is detected, the receiver of the packet sends a retransmission request to the transmitter. Packets that are received correctly may be, implicitly or explicitly, acknowledged as well.

Step 4: If the packet was received correctly, the receiver stores the received packets in its retransmission FIFO. Depending on the identifying information in the packet, the receiver will insert any received packet in the appropriate location in the retransmission buffer. The size of the buffer should be large enough to buffer packets until a retransmission of an initially corrupted packet is received. This is a function of the roundtrip delay of the transmission path, which is the time between the transmission of the packet and reception of the acknowledgement.

3.2.1 Pros and Cons of Retransmission

Retransmission offers the advantage that the overhead for correction of packets, which are affected by impulse noise, is only used when impulse noise actually occurs[2]. In the absence of impulse noise, retransmission does not incur an overhead penalty and can operate at its full data rate. One drawback of retransmission is the inherent jitter in data rate. This means that the incoming data stream is interrupted whenever a packet needs to be retransmitted to correct a packet that was not correctly received on the first attempt. Also, the maximum data rate that can be achieved by the system depends on the size of transmit and receive buffers (or, conversely, the maximum data rate will impose a certain minimum buffer size).

[2] Note that there is a fixed overhead associated with the required structure of the packets, including the identifying information and the error-detection redundancy.

4 G.998.4: Improved Impulse Noise Protection for DSL Transceivers

4.1 Introduction

The demand for improved impulse noise protection that is driven by new services was recognized by the standardization body ITU-T. In June 2007, ITU-T committee SQ15/Q4 (which previously developed the successful family of ADSL and VDSL standards) agreed to commence work on a new recommendation to “address improved impulse noise protection based on defined requirements primarily driven by IPTV”. The new work was informally referred to as **G.inp**. G.inp in itself is not a stand-alone standard, but needs to be implemented in conjunction with G.992.3 (ADSL2), G.992.5 (ADSL2+), or G.993.2 (VDSL2). The main technical solution contained in G.inp is a variant of the generic retransmission scheme illustrated in **Figure 3**.

After consent of the Recommendation in October 2009, the document received the official ITU number G.998.4. The publication of the final G.998.4 Recommendation is expected to occur in the first or second quarter of 2010.

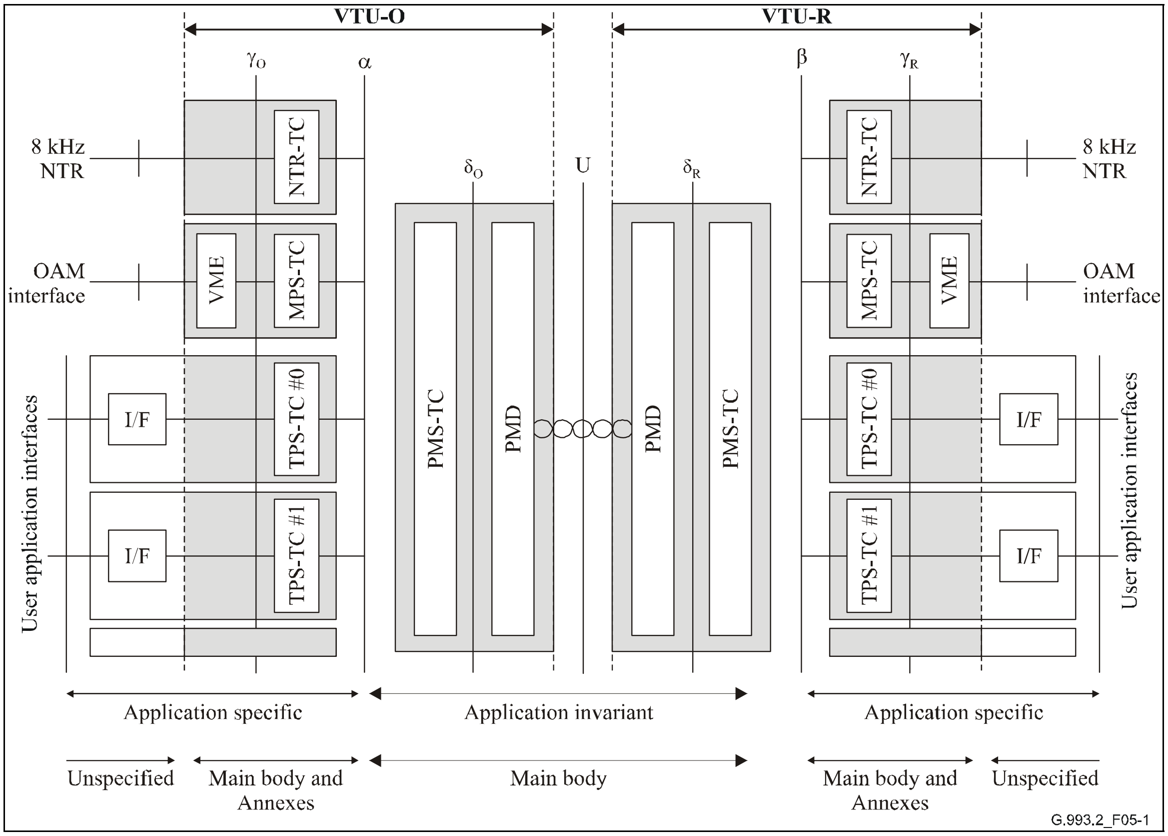
4.2 Retransmission Schemes

Initially, two independent retransmission schemes were considered based on the construction of data packets (as shown in **Figure 3**).

- **Scheme 1:** In this scheme, data packets were constructed above the γ-interface (see **Figure 4**) making the packets and the retransmission operation entirely transparent to the underlying DSL recommendation—see **Figure 4**. This approach is similar to the approach followed in bonding. In this case, as with bonding, the physical layer recommendation (ADSL2 or VDSL2) is agnostic to the operation of retransmission.
- **Scheme 2:** In this scheme the retransmission function is integrated inside the physical layer recommendation by defining a retransmission packet at the α/β interface (see **Figure 4**) as an integer number of RS codewords.

Ultimately, both schemes were combined into a final unified scheme, where management of retransmission queues can be implemented transparently in either the **PMS-TC** or in the **TPS-TC** layers of the DSL reference model—see **Figure 4**. Modems are unaware of the layer that the other modem has implemented the retransmission buffers.

Figure 4: Reference Model for DSL Transceiver: Interfaces and Functional Layers



4.3 DTU: The Retransmission Data Unit

G.998.4 defines a retransmission method for both ATM and packet-based (PTM) transmission protocols. For the purpose of retransmission, a new transmission unit called DTU (Data Transfer Unit) is created. In the **Retransmission Schemes** listed above, one of the objectives of G.998.4 was to accommodate systems that implement the retransmission buffers above the γ-interface, as well as systems that implement the retransmission buffers below the α/β-interface—as shown in **Figure 4**. To achieve this objective, a DTU is defined such that the DTU payload contains an integer number of protocol units (either ATM cells or PTM fragments). At the same time, the DTU is designed such that it consists of an integer number of Reed-Solomon codewords, as shown in **Figure 5**.

Figure 5: Example of DTU Structure

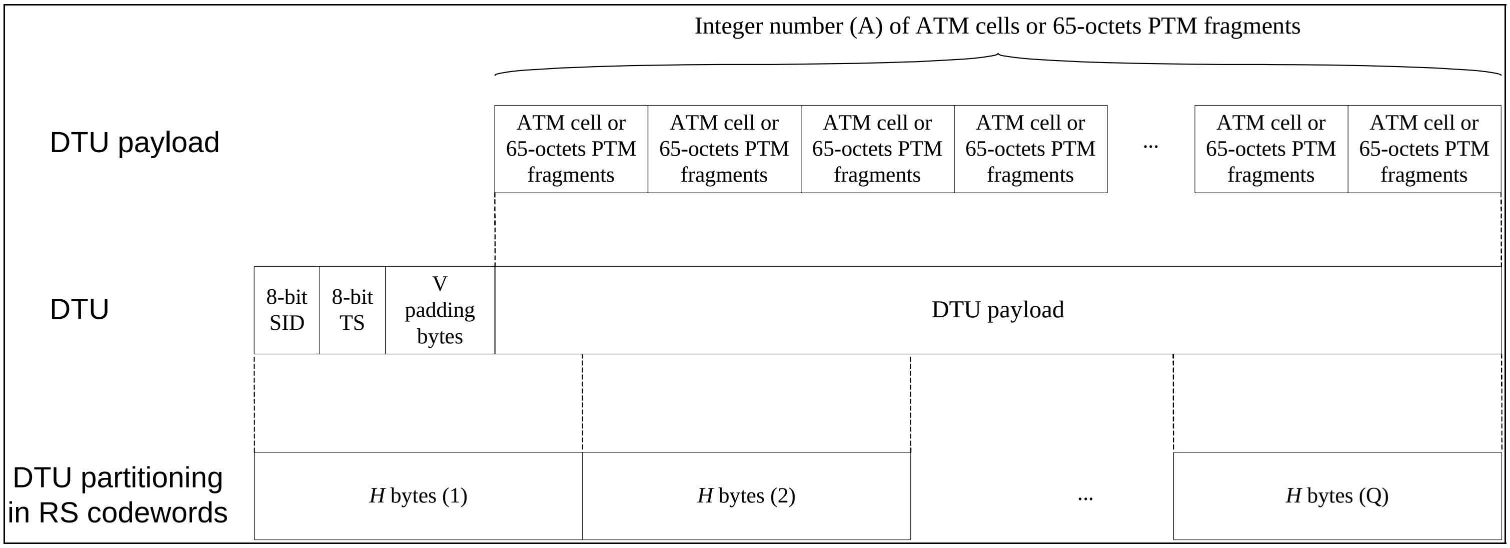


Figure 5 details the various elements of a DTU, where:

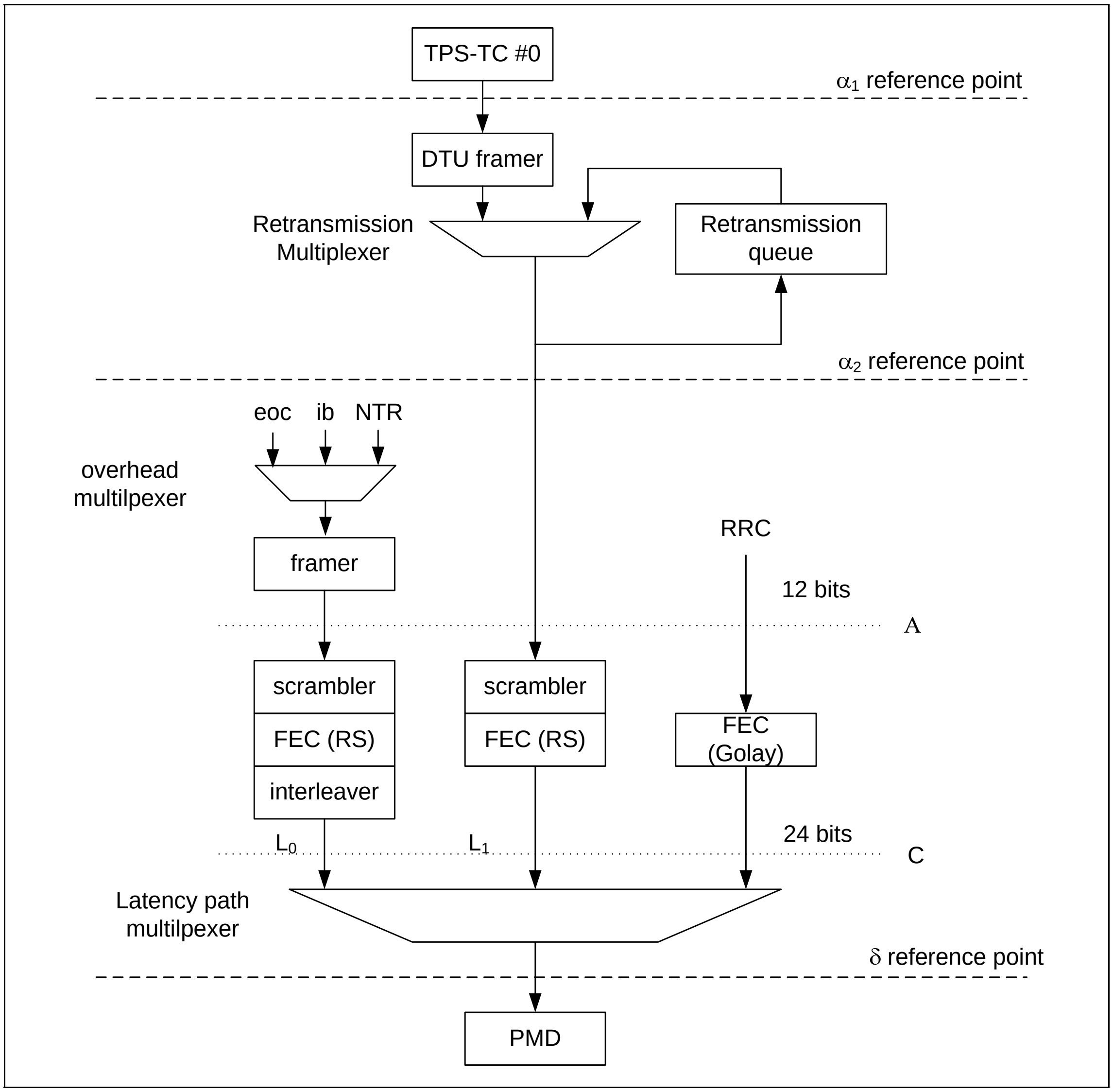
- The DTU payload consists of an integer number (A) of protocol units. A protocol unit is either a 53-byte ATM cell or a 65-byte PTM fragment, depending on the e of system.
- To form the DTU, the DTU payload is extended with various types of DTU overhead:
  - An 8-bit Sequence Identifier (SID) is added to allow transmitter and receiver to identify specific DTUs.
  - An 8-bit Time Stamp (TS) that records the time a DTU is first transmitted. The time stamp is not modified when a DTU is retransmitted. It is used to monitor whether a DTU complies with the delay bounds imposed by the system operator. If a DTU exceeds the maximum allowed delay, it will no longer be retransmitted.
  - A variable number (V) of padding bytes to allow the system flexibility in constructing the DTU and satisfying the various framing constraints.
- The DTU, consisting of DTU payload and DTU overhead, is constructed such that it can be partitioned into an integer number (Q) of Reed-Solomon codewords (before encoding).

In total, G.998.4 defines four framing types for the DTU. The DTU structure shown in Figure 5 is known as “Framing Type 1”.

Framing Types 2 to 4 are similar to framing type 1, but they also contain a CRC calculated over the DTU as part of the DTU overhead, to facilitate detection of DTUs with processing above the γ-interface.

4.4 Retransmission Reference Model

Figure 6: G.998.4 Reference Model (Including Retransmission)



In **Figure 6**, the retransmission queue is shown at the α-interface. This buffer and the DTU framer function can also be located above the γ-interface without disturbing the reference model.

Figure 6 illustrates the G.998.4 reference model showing the sequence of retransmission:

1. A DSL modem that implements retransmission in the forward transmission direction supports a single bearer channel (TPS-TC #0).
2. The DTU framer extracts data from the bearer channel to form DTUs according the desired DTU framing type. **Figure 5** shows an example of Framing Type 1.
3. The DTUs are placed in a Retransmission Queue after the DTU has been transferred over the α_2 interface —that is, when it has been forwarded for further processing to be transmitted

over the channel. Instead of transferring a new DTU, the retransmission multiplexer may decide to retransmit an older DTU from the retransmission queue.

4. Any DTU that crosses the α_2 interface is further processed for transmission over the channel. Firstly, the DTU is scrambled and encoded using Reed-Solomon coding. Subsequently, the bytes are mapped into a dedicated latency path (latency path #1).
5. DSL overhead traffic, such as eoc, indicator bits (IB) or Network Timing Reference (NTR), is separately encoded and framed before being mapped into latency path #0.
   As part of the G.998.4 agreed reference model, latency path #0 carries only overhead traffic, while latency path #1 carries only encoded user data.
   This is a constraint that is permitted within the underlying DSL recommendations (G.992.3, G.992.5 and G.993.2), but is mandatory when those recommendations are used in conjunction with G.998.4.
6. In addition to the two latency paths, containing overhead and user data respectively, each DMT symbol also contains 24 bits that carry the Retransmission Request Channel (RRC). The RRC transfers information regarding the status of received DTUs and allows the other side to determine DTUs that require to be acknowledged and DTUs that must be retransmitted. Note that the RRC is only present if retransmission is enabled in the reverse direction. The RRC path data is encoded with a variant of the well-known Golay code.

For VDSL2 (G.993.2), retransmission can be enabled for both upstream and downstream transmission. For ADSL2 (G.992.3) and ADSL2+ (G.992.5), retransmission is only defined in the downstream direction for error detection.

4.5 Retransmission Configuration and Reporting

In Legacy INP schemes, (as described in Legacy Impulse Noise Mitigation Scheme: FEC and Interleaving) the desired level of protection against impulse noise is configured using a series of management information base (MIB) parameters. Most importantly, the operator must specify the impulse length for which protection is required, as well as the maximum delay that the system may occur in selecting the configuration.

Similarly, if the modems are configured to operate with retransmission enabled, the desired impulse noise protection with retransmission is configured using a series of MIB parameters through a configuration interface. This configuration interface replaces the legacy configuration interface. MIB parameters include:

- The SHINE impulse length for which protection is required (INPMIN_SHINE_RTX)
- The REIN impulse length for which protection is required (INPMIN_REIN_RTX)
- The assumed frequency of the REIN impulses (either 100 or 120 Hz) (IAT_REIN_RTX)
- The minimum and maximum allowed delay (DELAYMIN_RTX and DELAYMAX_RTX)
- The maximum overhead due to retransmissions of SHINE impulses (SHINERATIO_RTX)
- Minimum and Maximum data rates

Based on the constraints imposed by the configuration, the receiver will determine the various framing parameters (including DTU structure and buffer sizes) to comply with the requirements. G.998.4 supports two types of impulse noise environment: one where all impulses are either of the REIN or SHINE impulses and one where the environment consists of a combination of REIN and SHINE impulses.

4.6 Using Retransmission with ADSL2, ADLS2+ and VDSL2 Equipment

The main body of G.998.4 defines the generic requirements needed for implementation of retransmission, such as the DTU structure, reference model and configuration parameters.

In addition, G.998.4 contains an Annex for each of the underlying recommendations (ADSL2, ADLS2+ and VDSL2). Implementation of retransmission requires compliance with the main body of G.998.4, one of the annexes and the corresponding recommendation. The annexes specify the requirements that are specific to each recommendation.

Support of retransmission mainly affects the initialization of the various recommendations since the initialization messages must be modified to enable modems to negotiate the use of retransmission.

5 Conclusion

This whitepaper introduces a new retransmission method that has been defined in the upcoming ITU-T standard G.998.4 (formerly known as G.inp).

To succeed in the triple-play market, telcos need to ensure that their DSL network is robust, reliable, and capable of delivering premium and enhanced services. Consumers have extremely high expectations for their QoE with no downtime, particularly with video and IPTV services.

Impulse noise imposes severe limitations to rate and reach performance of telcos' xDSL systems. With **Ikanos' Retransmission Technology**, telcos can be confident of delivering a stable, guaranteed, always-on solution with predictable Quality of Service (QoS), particularly in an impulsive noise environment. This capability is critical for a successful deployment of real-time triple-play and IPTV applications. xDSL chipsets must support standard-based retransmission capabilities, such as G.988.4, to provide robustness against impulse noise. To sustain this demand, G.988.4 retransmission scheme that is fully integrated with the available **Ikanos innovative iQV™ technology**, will provide unsurpassed service quality and reliability in an end-to-end, IP-based access network.

Thus, a unified and integrated 'reliability' solution, such as Ikanos iQV™, enable telcos to maintain–and more importantly, exceed–the Quality of Service and user experience their customers have come to expect.

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