0.1 ISDN - Integrated Services Digital Network

ISDN - Integrated Services Digital Network is a telephone system network which was designed to support a wide range of services that would appear to the user as being provided by an integrated network. The key feature of ISDN is that it was designed to be implemented with the existing subscriber loop copper cables used by the plain old telephone system (POTS).

The transmission and switching parts of the public switched telephone network (PSTN) had been mostly digitized before ISDN was designed. Only the subscriber loop communication and signaling remained analog. The spectrum between 300 Hz and 3400 Hz was allocated to voice and data transfer communications. A modem was used to modulate the digital signal into an analog one, which would then be transported within the voice channel. The quality of the transmission and the small bandwidth used by PSTN limited the data rate at a speed of 56 kbps. The demand for a network solution that could also provide Internet access, video telephony, and video conferencing, along with the basic telephony service, was soon noticed among subscribers. ISDN arose as a circuit-switched [1] telephone system which could also provides access to packet switched networks at considerably higher rates.

Although ISDN has been actively researched since the 1970s, and a first set of ITU-T ISDN Recommendations was published in 1984, it was far from being complete. It laid down a set of general concepts that allowed manufacturers in the early days to proceed with the design of the first generation of ISDN equipment. In 1988, the ITU-T published a more elaborate standardization of the ISDN under the name of the Blue Book Recommandations, but by that time, several nations embarked on the definition and deployment of their own national standards which made it impossible to have a single common worldwide standard.

0.1.1 Data rates

The performance of digital transmission on the already installed copper cable in the subscriber loops determined the ISDN data rates. The Basic Rate Access (BRA) ISDN, uses two bi-directional B channels, also called bearer channels, for data and voice transmissions, and one D channel for call control signalling. Each of the two B channels has a data rate of 64 kbps. The D channel has a data rate of 16 kbps. The basic rate access is also referred to as 2B+D, and offers a total data rate of 128 kbps to the user. The B channel data rate comes from the fact that the highest frequency considered for sampling and digitization purposes in PSTN is 4 kHz. The Nyquist sampling criterion dictates that the sampling rate should be at least twice the highest frequency, which gives a sampling rate of 8 kHz. Each sample is then assigned an eight-bit pulse code modulation (PCM) symbol. In this way, a channel data rate of 64 kbps (8 kHz $\cdot 8$) is achieved.

[1] A circuit-switched network establishes a fixed circuit for the duration of a call between the end points as if they were physically connected by a wire. Once the call is terminated, the circuit may be switched to a new connection. In the case of a packet switched network, the data traffic is split into packets which are routed independently to the destination.

A third type of channel defined by ISDN standards is the H channel, which is a higher bandwidth channel used for high speed data transfer, such as video conferencing or Local Area Network (LAN) interconnection. There are currently three types of H channels: H0, H11 and H12. The first type operates at a data rate of 384 kbps (six times the data rate of a B channel), the second operates at 1.536 Mbps, and the third at 1.920 Mbps. All the three channel types B, D and H are multiplexed in different ways to achieve higher data rates.

The ISDN standard also defines a Primary Rate Access (PRA) interface which reaches data rates of 1.544 Mbps for North America and Japan, and 2.048 Mbps for Europe. For a B-channel PRA structure, 23 B channels and a 64 kbps D channel are multiplexed for North America and Japan, and 30B+D channels for Europe. In the case of an H-channel PRA, some possible interface structures are 4H0 or 3H0+D at 1.544 Mbps for North America and Japan, and 5H0+D at 2.048 Mbps for Europe. There is also the possibility of having the D channel provided by another interface within the same access, and in that case, a rate of 1.544 Mbps would be achieved by a single H11 channel, and a rate of 2.048 Mbps by a single H12 channel. Due to crosstalk, PRA transfer rates could only be achieved using the standard analog subscriber loop cables by fulfilling separation conditions.

0.1.2 Subscriber loop reference model

The ITU-T Recommendation I.441 defines the types of equipment used, the cabling interfaces within the customer premises and the possible ways of interconnecting and configuring the equipment. Figure 1 shows the reference model of the ISDN subscriber loop:

TE1 is defined as a terminal equipment of type 1. Type 1 equipment can establish and terminate a call without the need of installing auxiliary equipment that would provide the interface and protocol processing for call setup. Examples of type 1 equipment are ISDN-ready devices such as ISDN telephones, ISDN fax machines, ISDN data/voice terminals and video phones.

TE2 is a terminal equipment of type 2, which does not have an ISDN user-network interface and requires a Terminal Adapter (TA) to interface to the ISDN network. A terminal adapter has compatible interfaces to both the non-ISDN equipment and the digital network and performs the protocol processing for call setup and termination for the TE2. Some other important functions of the TA include rate adaptation and multiplexing. TE2 devices that require data rates lower than the 64 kbps are also allocated a full channel, and the bit rate is achieved by insertion of padding bits. This is a waste of bandwidth and due to this reason, the TA was designed to multiplex several bit streams into a single ISDN bearer channel. Examples of equipment that requires a terminal adapter are PC's, routers, modems, and the POTS telephones. In the case of a PC, the TA is the plug-in card and the R interface is the machine's system bus.

NT1 is a network termination of type 1. It provides a conversion at the physical layer between the S interface cable, that runs inside customer premises, and the subscriber loop cable at the U interface, that connects to the local exchange.

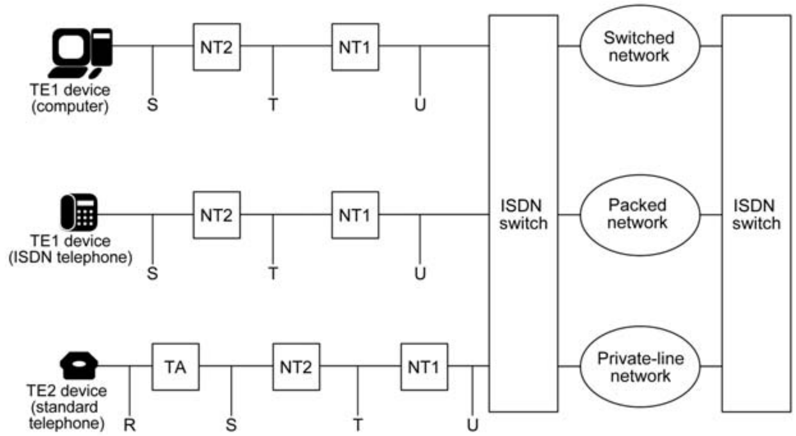

Figure 1: ISDN Subscriber loop reference model

NT1 equipment is responsible for the conversion of the central office 2-wire, echo cancelled, 2B1Q code to a customer premise 4-wire, pseudo-ternary, encoding. Power feeding of the terminal from local mains supply or from the network, as a backup when mains power fails, is also one of the features of NT1.

NT2 is a network termination of type 2, which provides functions such as switching or multiplexing. In some customer premises, such as the case of company offices, the calls are internally routed by a Private Branch Exchange (PBX). A PBX provides the users with internal voice services, as well as external services. In some cases, the NT1 equipment is integrated into the PBX.

The end of the subscriber loop is marked by a line termination (LT) which is found in the local exchange. A local exchange is responsible for providing control over the local subscriber loops during calls. It also provides signalling capabilities to other exchanges for call setup and termination, as well as access to packet-switched networks.

The ISDN Recommendations define four interfaces between the network equipment, as seen in Fig. 1. The R interface between a TE2 device and a terminal adapter, the S interface between a TE1 device and a NT2, the T interface between the NT1 and NT2 and the U interface on the network side, between the customer premises and the local exchange. For the case in which NT2 is built into NT1, the S interface is referred to as the S/T interface.

The S/T interface

As previously mentioned, the S/T interface connects the TE and NT equipment and provides two modes of operation - point-to-point and point-to-multipoint. The point-to-point and two most common point-to-multipoint configurations are presented in Fig. 2.

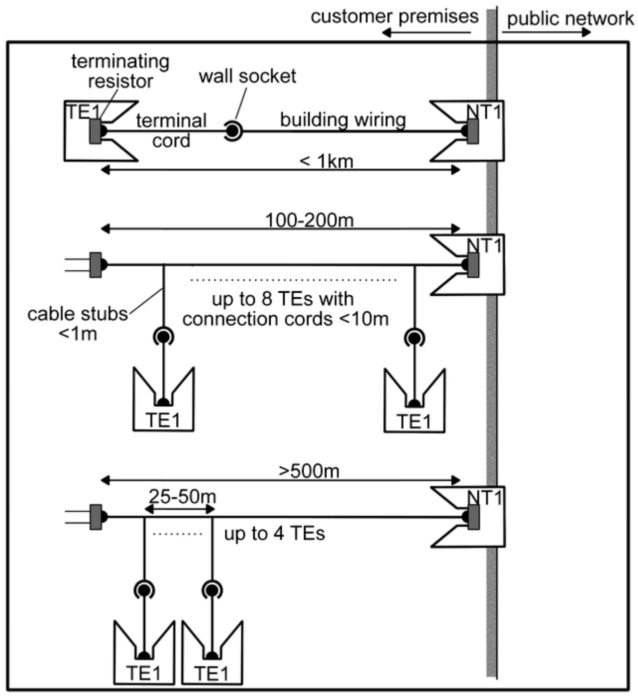

Figure 2: Wiring configurations a) Point-to-point b) short passive bus c) extended passive bus

In the case of a point-to-multipoint access mode, a maximum of eight TEs can be connected to the network. The B channels are allocated on a call-by-call basis and the D channel is shared among all the TEs connected to the S bus cable. A collision detection mechanism for the D channel has been provided, that echoes back to the TEs the value of the previous D channel bit that was received by the network. The TEs receive the echo bits and compare them to what they sent and, in case of mismatches, the transmission is stopped in the D channel. The full duplex in the case of the S transmission system is realized by the use of two pairs of wires. One of them conveys information in the TE-NT direction, and the other one in the NT-TE direction. The advantage of using two pairs of wires instead of a single one lies in the fact that there is no need for an echo canceller, which simplifies the transceiver circuitry. The line termination for the S transmission system is 100 Ω.

Once the S transmission system has been activated, frames are transferred between the TE and the NT. In the case of a BRA structure, a rate of 192 kbps is achieved, two 64 kbps bearer (B) channels, one 16 kbps D channel and 48 kbps additional framing and maintenance channels that are transparent to the user and are only present within the Layer 1 of the OSI reference model. Alternate Mark Inversion (AMI) is used to encode the frames for a better transmission quality. For a binary 1, no line signal is transmitted and for a binary 0, alternating polarity signal levels of plus or minus 750 mV. In order to avoid charge accumulation on the lines, a zero DC balance should be achieved. This is done by splitting the frame into sections which are individually balanced by the inclusion of a DC balance bit.

The beginning of a new frame is signaled by the violation of the coding rule. For instance, for two consecutive binary zeros, alternating high/low signal levels would be expected, but instead, the signal level is not changed for the second bit. Although all frames have a standard size of 48 bits as can be seen in Fig. 3, the structure is different for the two directions of transmission - from the TE to the network or from the network to the TE.

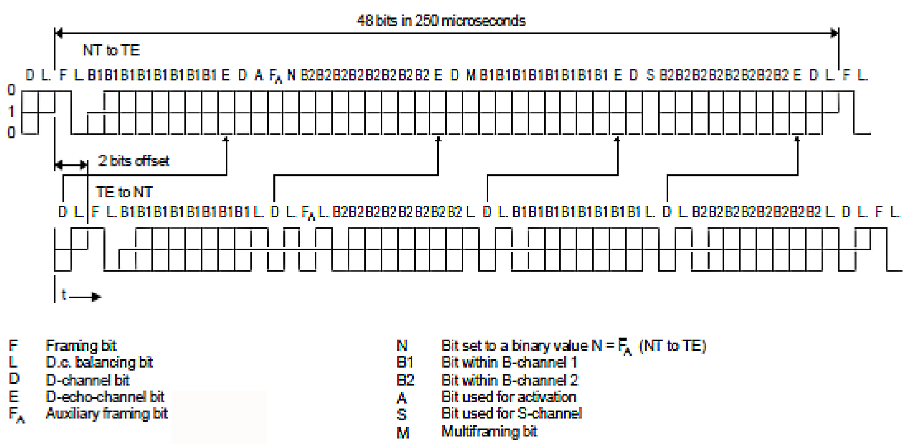

Figure 3: I.430 – Frame structure

Remote Power Feeding

Remote feeding across the ISDN subscriber loop allows the delivery of power to both NT and TE. In case of mains supply failure, network operators are required to provide basic life-line services to the users. Normally, the NT can be fed from the network or from a local mains supply or battery. A common way of supplying the TE with power is by using a phantom circuit such as the first one in Fig. 4. This circuit uses four wires and superimposes a DC supply voltage on the transmitted signals by means of center-tapped transformers. This type of remote feeding circuit has its limitations when it comes to the amount of power that can be delivered to a terminal. Another way of remotely feeding the terminal is by using a separate pair of wires in the S cable that is not used for transmission. The advantage is that this scheme is capable of delivering more power to the terminal than the phantom circuit. There is also the possibility of having one TE provide power to other TEs with which it shares the same

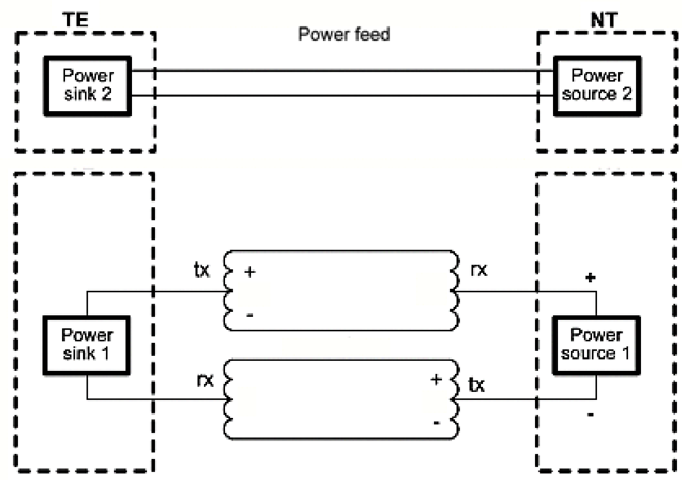

Figure 4: Remote feeding

multidrop bus. The ISDN standard defines two states in which power can be remotely provided to the TEs – the normal and the restricted state. For emergency situations the restricted mode is activated and one TE is elected as the designated one from all the TEs that share the same bus.

The U Interface

Unlike the S transmission system, the BRA U transmission system, which connects the NT at the user premises to the local exchange (LE) uses two wires instead of four, the same twisted pair wires used previously for analog subscriber loops. The U cable might be composed of a single continuous length, or of several interconnected sections with an overall length of up to 8 km, and might also include bridged-taps. A hybrid circuit is used to multiplex/demultiplex the received and transmitted signals within the NT as to switch from two wires to four wires. The hybrid circuit is also referred to as a two-to-four wire converter.

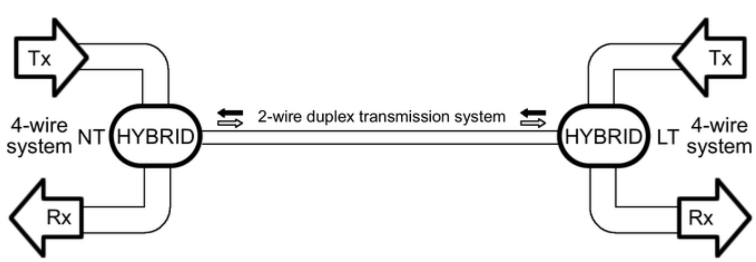

Figure 5: A typical 2-wire transmission system using a hybrid circuit

Although ISDN uses the same spectrum for upstream and downstream and a single pair of wires for the U system, full duplex is achieved by making use of echo cancellers to distinguish between the received and transmitted signals. The cancellation works by subtracting a delayed version of the transmitted signal – the echo – from the signal that was received. For a typical BER performance of 10^{-7}, the echo must be suppressed by 60 dB. The typical 2-wire duplex transmission system is illustrated in Fig. 5 and the functions of a U transceiver are illustrated in Fig. 6.

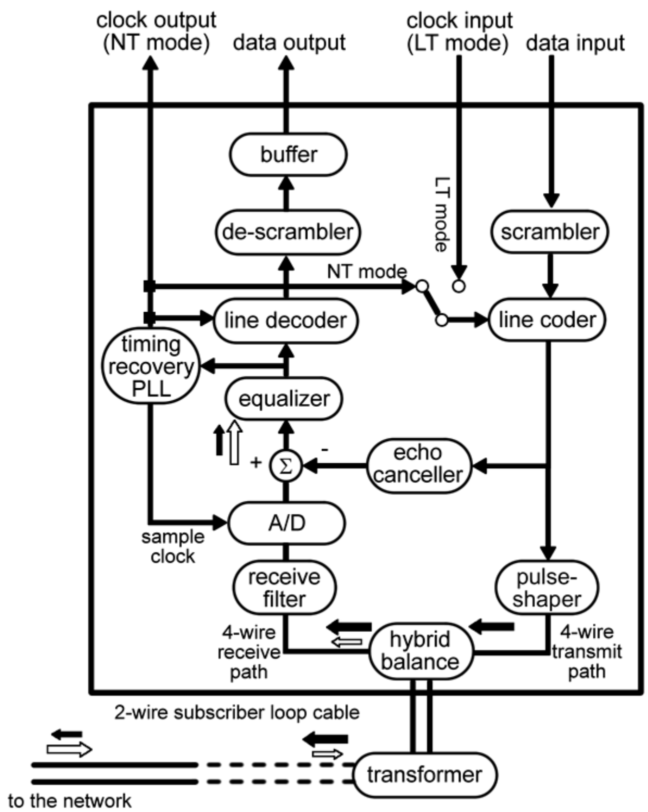

Figure 6: U transceiver - main functions

The U transceiver performs scrambling, line coding, and pulse shaping before sending data through the cable. Although the received symbols represent digital data, the presence of an analog-to-digital (A/D) converter in the received path is explained by the fact that the pulses get distorted due to the transmission cable properties before reaching the far-end transceiver. Even if the pulse spreading effect is also present in the S transmission system, its effects are far more severe in the U cable mainly because of the bigger length of the cable. Decision feedback equalizers (DFEs) are used to cancel the effect of the signal dispersion. Both equalization and echo cancellation are required before the slicer[2] makes a decision on the value of the received signal. Before sending it out, data is descrambled and buffered.

[2]Included in the line decoder

When implementing such a system, one has to consider the possible noise sources present and the characteristics of the subscriber loop cable. Self-NEXT is one of the major disturbances of the U system. It results from the interference of wires within the same system that are using the same spectrum for both upstream and downstream transmission, as in the case of ISDN. Far end crosstalk (FEXT) is highly attenuated due to the transmission length of the cable and drops below the noise level, which does not make it a major disturber. Coupling among adjacent wires is a power-function of frequency and can be minimized by using a line code with a low bandwidth. Impulse noise is normally generated by network equipment transients and light switching and, in most cases, most of its energy is contained below 50 kHz. A reasonable solution would be to place a high pass filter before the echo canceller in the receive path. The use of error correcting codes is also a common practice to recover the information affected by the impulse noise.

Regenerators (REGs) are inserted in the loop between NT and LT to recover the signal and bridge distances larger than 4.2 km. Nevertheless, a regenerated link is expected to fit within the thresholds for bit-error ratio (BER) and delay which are defined in the same way for the non-regenerated links.

0.1.3 Line coding

2B1Q

As mentioned earlier, a low bandwidth code is required in order to keep the coupling among wires at a minimum, and to ensure a lower attenuation of the signal energy as it travels through the network. A line code that would match the above requirements was chosen – 2B1Q (2 Binary, 1 Quaternary)[3]. This code works by encoding every two bits of data as one of the four equally spaced signal levels, doubling the period of each symbol, and achieving a baud rate of 80 kbaud/s. This places the bandwidth of the 2B1Q system in the lower region on a power spectral density graph. Since telephone transmission lines act as a low-pass filter, a low bandwidth system will achieve greater reach.

The disadvantage of a four-level code such as this one is that the reduced spacing between the levels results in a higher BER for the same average power. However, for longer cables the attenuation at high frequencies has a more severe effect on the transmission quality than a reduced signal-to-noise ratio due to the use of a four-level code, so the benefits of using 2B1Q overcome the drawbacks.

2B1Q encoded frames are used to transfer information between NT and LT. The basic 2B1Q frame is 120 symbols (240 bits) long. The first 18 bits represent a synchronization word, while the following 216 bits carry 2B+D channels data, and the remaining 6 bits are used for maintenance operations. The standard duration of a 2B1Q frame is 1.5 ms. Eight basic frames are combined into a single multiframe, or superframe, and each superframe has a time duration of 12 ms. Figure 7 presents the superframe structure.

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[3] 2B1Q = 4–PAM

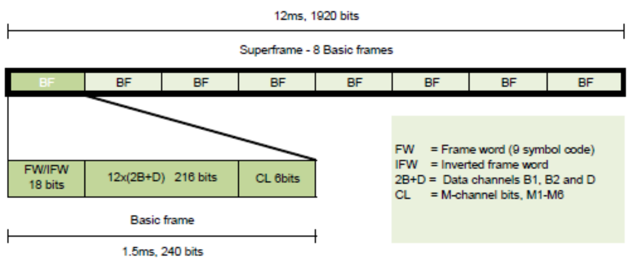

Figure 7: ETR 080 – 2B1Q superframe

4B3T (MMS43)

Another encoding scheme primarily used in Germany is MMS43, which is derived from 4B3T (Four Binary Three Ternary). This block code represents a symbol of four bits by three pulses which are looked up in a table, each pulse taking one of three values. This means there are 16 (2^4) input symbols and 27 (3^3) output combinations. MMS43 achieves an overall zero DC bias by using four different code sets and switching between them. The current DC level is monitored and for the next symbol to be encoded, a code from another code set is chosen such that the DC offset is reversed. Table 1 presents the MMS43 code sets along with the number of the next code set to be used. MMS43 decoders benefit from a reduced complexity due to the fact that they are not required to keep track of the DC offset. Unlike 2B1Q which has a baud rate of 80 kbaud/s, MMS43 has a baud rate reduction from 160 kbit/s to 120 ($160 \cdot 3/4$) kbaud/s, only. A higher reduction in the baud rate, such as the one of 2B1Q, results in a higher spectral efficiency which, in turn, leads to reduced line attenuation and improved immunity to noise and crosstalk.

Table 1: MMS43 encoding – the four code sets

Input symbols	1	2	3	4
0001	0 − + 1	0 − + 2	0 − + 3	0 − + 4
0111	− 0 + 1	− 0 + 2	− 0 + 3	− 0 + 4
0100	− + 0 1	− + 0 2	− + 0 3	− + 0 4
0010	+ − 0 1	+ − 0 2	+ 0 − 3	+ 0 − 4
1011	+ 0 − 1	+ 0 − 2	+ 0 − 3	+ 0 − 4
1110	0 + − 1	0 + − 2	0 + − 3	0 + − 4
1001	+ − + 2	+ − + 3	+ − + 4	− − − 1
0011	0 0 + 2	0 0 + 3	0 0 + 4	− − 0 2
1101	0 + 0 2	0 + 0 3	0 + 0 4	− 0 − 2
1000	+ 0 0 2	+ 0 0 3	+ 0 0 4	0 − − 2
0110	− + + 2	− + + 3	− − + 2	− − + 3
1010	+ + − 2	+ + − 3	+ − − 2	+ − − 3
1111	+ + 0 3	0 0 − 1	0 0 − 2	0 0 − 3
0000	+ 0 + 3	0 − 0 1	0 − 0 2	0 − 0 3
0101	0 + + 3	− 0 0 1	− 0 0 2	− 0 0 3
1100	+ + + 4	− + − 1	− + − 2	− + − 3

0.1.4 Spectra

According to [2], the single-sided power spectral density[4] of an 80 kbaud 2B1Q signal with full-baud square-topped pulses and with 2nd order Butterworth filtering ($f_{3db} = 80$ kHz) is given by:

$$PSD_{ISDN}(f) = \frac{5}{9} \cdot \frac{{V_p}^2}{R} \cdot \frac{\left[\sin\left(\frac{\pi \cdot f}{f_0}\right)\right]^2}{\left(\frac{\pi f}{f_0}\right)^2} \cdot \frac{1}{1 + \left(\frac{1}{f_{3db}}\right)^4} \tag{1}$$

Where $f_0 = 80$ kHz, $V_p = 2.5$ V and $R = 135\ \Omega$. [7] provides a formula for the MMS43 power spectral density:

$$PSD_{MMS43}(x) = \frac{1}{624}[479 - 48 \cdot \cos(x) + 16 \cdot \cos(2x) - \frac{v_0 + v_1 \cdot \cos(x) + \ldots + v_5 \cdot \cos(5x)}{n_0 + n_3 \cdot \cos(3x) + n_6 \cdot (6x)}]$$

Where

$$\begin{aligned} v_o &= 301339 & n_0 &= 4505 \\ v_1 &= 426304 & n_3 &= -2440 \\ v_2 &= 171688 & n_6 &= -384 \\ v_3 &= -62432 & x &= 2\pi f T_C \\ v_4 &= -62352 \\ v_5 &= -23120 \end{aligned}$$

See Fig. 8 for the PSD of an 80kbaud 2B1Q signal with random equiprobable levels, full-baud square-topped pulses and a second order Butterworth filtering. In the case of MMS43, only a rectangular window was used.

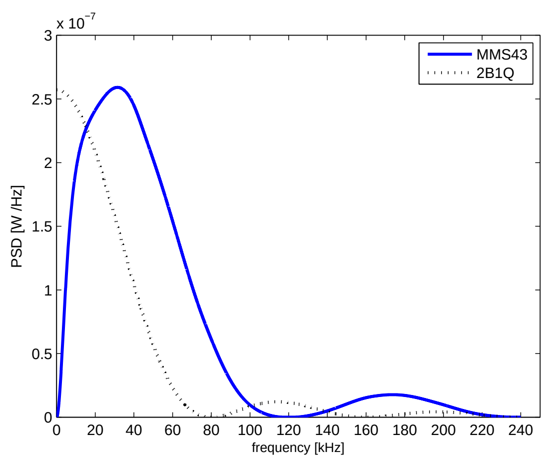

Figure 8: 2B1Q and MMS43 power spectrum densities, ISDN BRA

[4]Power Spectral Density (PSD) is provided as a ratio of the power in one Hertz of bandwidth, where power is expressed in units of dBm.

The following characteristics apply to the 2B1Q U transmission system (NT1, REG and LT), with a line impedance of 135 Ω. Figure 9 shows the upper bound of the power spectral density of the transmitted signal. The transmitted pulses should fit within the mask showed in Fig. 10. According to [6],the quaternary symbols are obtained by multiplying the pulse mask with +2.5 V, 5/6 V, −5/6 V or −2.5 V.

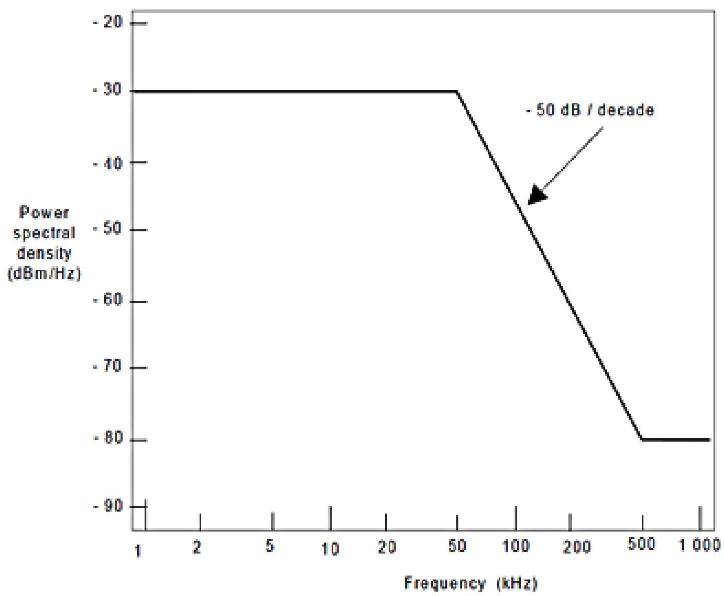

Figure 9: ETR 080 – 2B1Q spectral mask

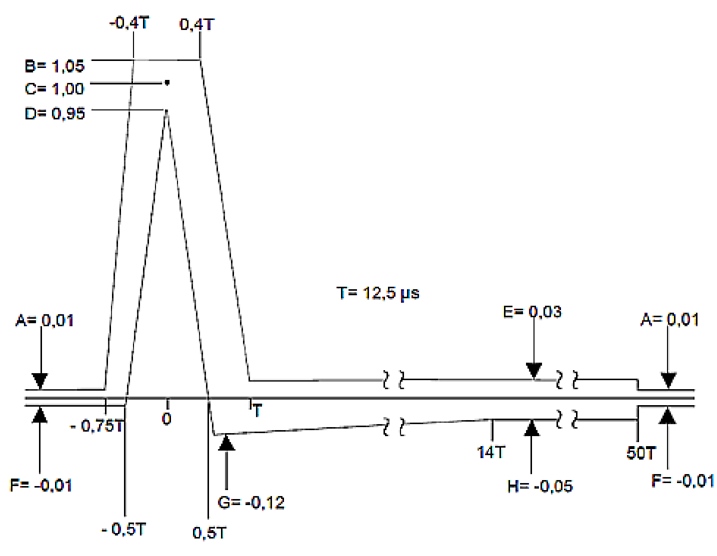

Figure 10: ETR 080 – Pulse shape for a 2B1Q transmission system

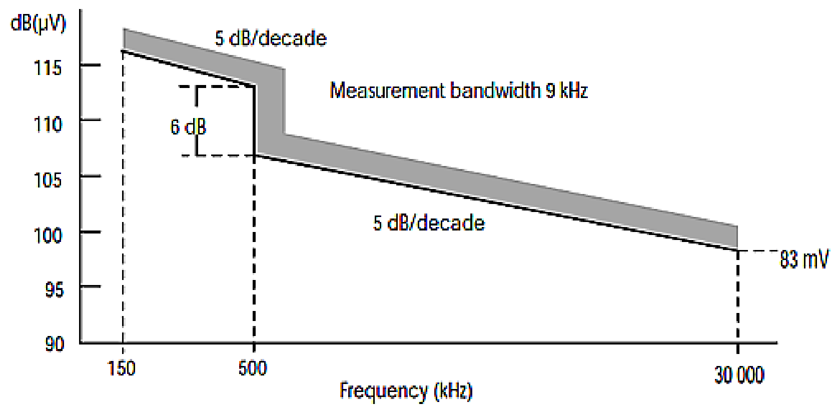

Figure 11: ETR 080 – MMS43 spectral mask

For the MMS43 U transmission system with a line impedance of 150 Ω, Fig. 11 presents the upper bound of the power spectral density. The shape of the transmitted pulse should fit within the mask in Fig. 12. The pulse amplitudes are defined as 2V, ±0.2V.

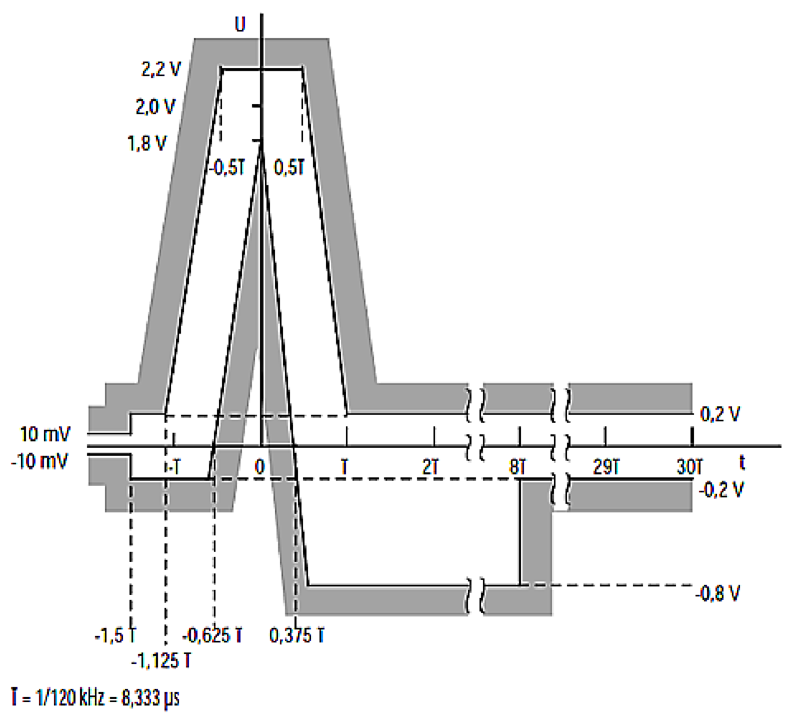

Figure 12: ETR 080 – Pulse mask for MMS43 U transmission system

0.1.5 Protocols and startup

Although ISDN standards have not been defined having the open systems integration (OSI) seven layer reference model in mind, this is considered a good place to start when looking for an abstract description of how ISDN works. The lower three layers of the OSI model (Physical, Data-link, Network) are used to describe the call control signaling across the user-network interface, which takes place within the D channel.

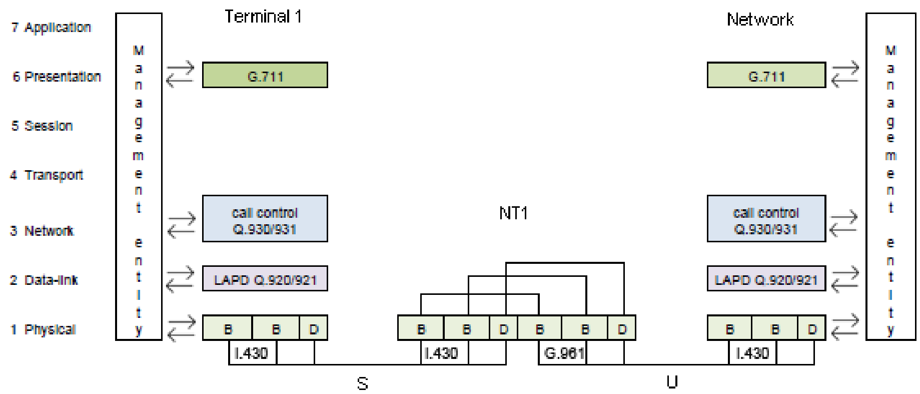

Figure 13: ISDN protocol stack

Layer 2 Protocol - LAPD

Link Access Pro - Channel D (LAPD) is a layer 2 protocol for the D channel, defined in Q.920/921 Recommendations, which, along with higher layer protocols, enables the handshaking, signaling, and control for voice and data calls. LAPD allows up to eight users to share the same D channel by assigning each of them a Logical Channel Number and bandwidth, on a first-come-first-served basis. LAPD is also responsible for framing, sequence control, and error detection.

LAPD frames consist of several fields as illustrated in Fig. 14. Each frame contains at most six fields – opening flag, address, control, information, frame check sequence, and closing flag. Certain frames do not contain the information field. The opening and closing flags mark the beginning and the end of a frame and are used for frame synchronization. The address field permits the existence of multiple logical data links by addressing the service access points (SAPI) and the terminal endpoints (TEI) independently. A terminal access point is an address which identifies a PPP[5] layer two connection which is established between a terminal and the network. The TEI is assigned by the network, following a terminal request, and is retained until the terminal is unplugged from the network or until a multiple assignment of the same TEI has been detected. The service access point identifier (SAPI) serves to identify the management entity that is associated with the data contained in the information field of a

[5]PPP – Point-to-Point Protocol is a data-link protocol used to establish a direct connection between two network end points. It also provides authentication, encryption and compression capabilities.

frame. Nevertheless, both TEI and SAPI identifiers do not go above layer two, and for this reason, every time a TEI is assigned to a terminal, a connection end-point identifier (CES) is allocated with it. This permits layer 3 services to be associated with a particular data-link connection.

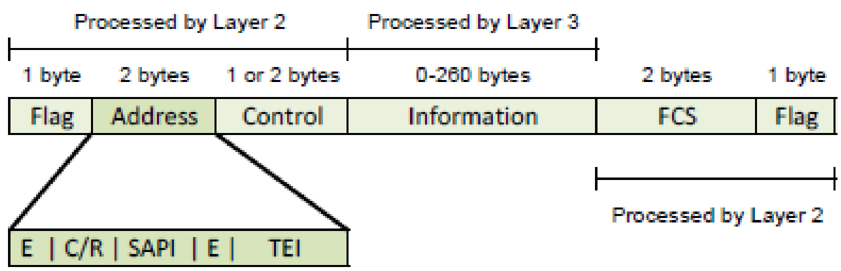

Figure 14: Q.920 LAPD frame structure

In addition to TEI and SAPI, the address field also includes a Command or Response (C/R) field. The control field specifies the type of frame being transmitted so that it can be processed accordingly. The Q. 921 Recommendation defines three types of frames – information (I), supervisory (S), and unnumbered (U). The information frames are used to transfer information among layer 3 point-to-point devices. The S frames are used for supervisory functions, such as acknowledgements of I frames and retransmission requests of I frames. The U-format defines unnumbered and unacknowledged frames that are used infrequently for sending data that does not require such reliable transmission.

The next field in the Q. 920 frame structure is the information field, which carries higher level data and whose size is limited to a maximum of 260 bytes. The frame check sequence (FCS) is a 2 bytes long cyclic redundancy checksum.

Layer 3 Protocol - Q.931

While the layer 2 LAPD protocol is concerned with the error-free transmission of information, the call control in the subscriber loop is performed by the layer 3 protocol defined by the Q.931 Recommendation. Call setup and management is performed by the network layer, along with the invocation of supplementary services. For a better understanding of the layer 3 functionality, this can be viewed as being subdivided into two sub-layers, one responsible for call control and the second one for protocol control. Every call is advancing through a sequence of logical states for the setup, connection, and disconnection phases in the protocol control sub-layer. The main purpose of call control is to provide processing functions that are application specific.

Q.931 distinguishes between various message categories - call establishment messages, call information phase messages, call clearing messages, and miscellaneous messages [5]. Table 2 presents the most important message types along with a short description of their use.

Table 2: Q.931 message types

Call establishment messages:	
ALERTING	sent to indicate that called user alerting has been initiated
CALL PROCEEDING	sent to indicate that the requested call establishment has been initiated and no more call establishment information will be accepted
CONNECT	sent by the called user to the network and by the network to the calling user, to indicate call acceptance by the called user
CONNECT ACKNOWLEDGE	sent by the network to the called user to indicate the user has been awarded the call. It may also be sent by the calling user to the network to allow symmetrical call control procedures
PROGRESS	sent by the user or the network to indicate the progress of a call in the event of interworking
SETUP	sent by the calling user to the network and by the network to the called user to initiate call establishment
SETUP ACKNOWLEDGE	sent to indicate that call establishment has been initiated, but additional information may be required
Call information phase messages:	
RESUME	sent by the user to request the network to resume a suspended call
RESUME ACKNOWLEDGE	sent by the network to the user to indicate completion of a request to resume a suspended call
RESUME REJECT	sent by the network to the user to indicate failure of a request to resume a suspended call
SUSPEND	sent by the user to request the network to suspend a call
SUSPEND ACKNOWLEDGE	sent by the network to the user to indicate completion of a request to suspend a call
SUSPEND REJECT	sent by the network to the user to indicate failure of a request to suspend a call
Call clearing messages:	
DISCONNECT	sent to clear an access connection
RELEASE	sent by the user or the network to indicate that the equipment sending the message has disconnected the channel
RELEASE COMPLETE	sent by the user or the network to indicate that the equipment sending the message has released the channel and call reference

Every Q.931 message has a protocol discriminator (PD), a call reference identifier (CR), and a message type identifier (MT). While these three fields are common to all layer three messages, an additional field for other information elements might be also present within some of them. The purpose of the protocol discriminator (PD) is to distinguish messages for user-network calls from other messages. The second part of the message, the call reference, identifies the call or registration/cancellation request at the local user-network interface to which the message applies.

The call reference identifiers are unique values assigned at the beginning of a call by the originating side. Once the call has been completed, the reference value is free to be reassigned to a later call. The message type field is used to identify the function of the message being sent. The last field of the message holds specific actions and attributes associated with a call.

8	7	6	5	4	3	2	1
Protocol Discriminator							
0	0	0	0	Call Reference			
0	Message Type						
Other information elements as required							

Figure 15: Q.931 message format

Start-up - Activation and deactivation

During the start-up process, a series of signals are produced by both NT and LT with the purpose of establishing the master-slave relationship. Synchronization of the receivers, training of the echo cancellers and equalizers takes places during this phase as well.

Activation and deactivation of the S system

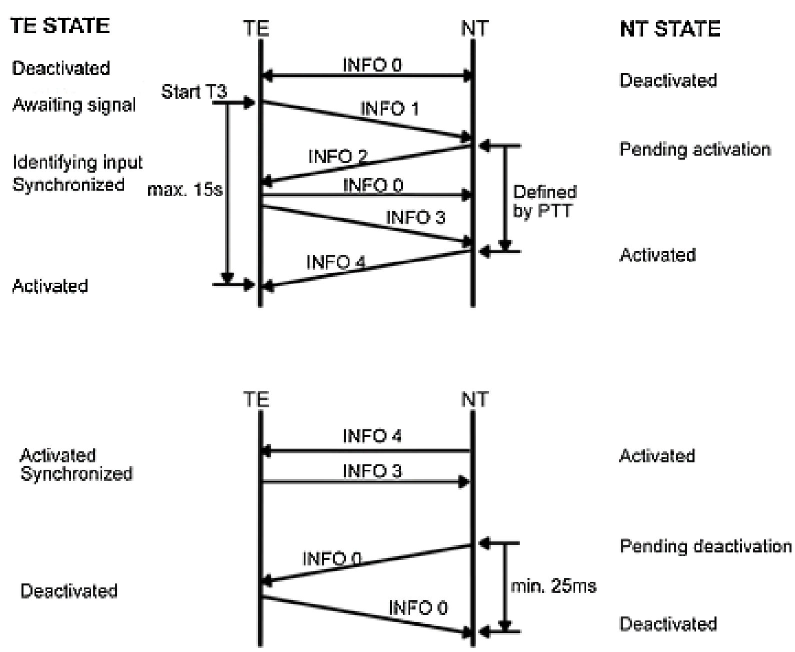

Figure 16: Activation and deactivation of the S transmission system

Prior to I.430 frame transmission, the S transmission system undergoes an activation phase that will permit receivers to be correctly synchronized with the transmitters. Both NT and TE make use of five different types of the so-called INFO signals. The NT derives its timing information from the 8 kHz master network clock and the TE makes use of the INFO signal received from the NT in order to derive the timing to synchronize its transmitted signal. The first type of signal, INFO 0 is used to indicate the absence of any active signal, all transceivers are fully deactivated. When a TE needs to establish a call, an INFO 1 signal is sent and the NT replies with an INFO 2 signal which permits the TE to lock onto the clock signal. Then the NT is signaled by the TE that it has achieved frame synchronization using an INFO 3 signal, and the NT ends the activation sequence by sending an INFO 4 signal. Figure 16 presents the activation and deactivation sequence diagrams for the S transmission system.

Activation and deactivation of the U transmission system

During the activation of the U transmission system, a sequence of signals is exchanged between the transceivers on the network, with the purpose of allowing the echo cancellers and the equalizers to train themselves and to synchronize the NT to the network clock. Activation at the U interface depends on where the call was initiated from. A cold start activation includes a training period during which the echo cancellers and the equalizers adapt to the transmission characteristics of the cable, and is normally performed when the transceivers on the network have just been turned on. Cold activation should not take more than 15 s, but even such a delay is considered to be uncomfortable during a call setup and this is why a faster activation is performed, referred to as warm start. When not in use, the equipment goes into a deactivated state, where the power consumption is minimized and when it recovers from this state it undergoes a warm start which uses the same equalizer coefficients that were computed during the cold start. This speculates the fact that the transfer functions of the channels do not vary much with time in wireline communications, as they do in wireless. A typical warm start activation is achieved in less than 300 ms. Figure 17 shows the cold activation sequence diagram initiated from the TE side. A call request is initiated by the TE by sending an INFO 1 signal. Before acknowledging with an INFO 2 signal, NT has to bring up the U interface and to synchronize itself with the network. It starts by sending a wake-up tone (TN) which serves to bring up the LT in case it had been previously deactivated or in a low power state. A SN1 signal follows, which consists of repeated basic frames. Once the NT1 has ceased transmission of SN1, indicated by SN0, the LT responds by transmitting signal SL1 and SL2. During this phase, when SN1 and SL1 are transmitted, the equalizers and echo cancellers are trained. SL2 signal allows the NT tranceiver to gain basic frame synchronization with the LT. This is acknowledged by a SN2 signal. Once this has been finished, NT proceeds further and sends an INFO 2 acknowledgment signal to the TE. At this moment in time, TE achieves synchronization with the network and informs NT by sending an INFO 3 signal. SN3 frames are used to announce the LT that full synchronization has been achieved. The NT announces the LT that full network synchronization was achieved and then LT signals the NT to complete the activation phase. The INFO 4 signal is used to inform the TE that the activation phase was completed. Deactivation can only be initiated from the network side when the U transceivers support warm-start activation.

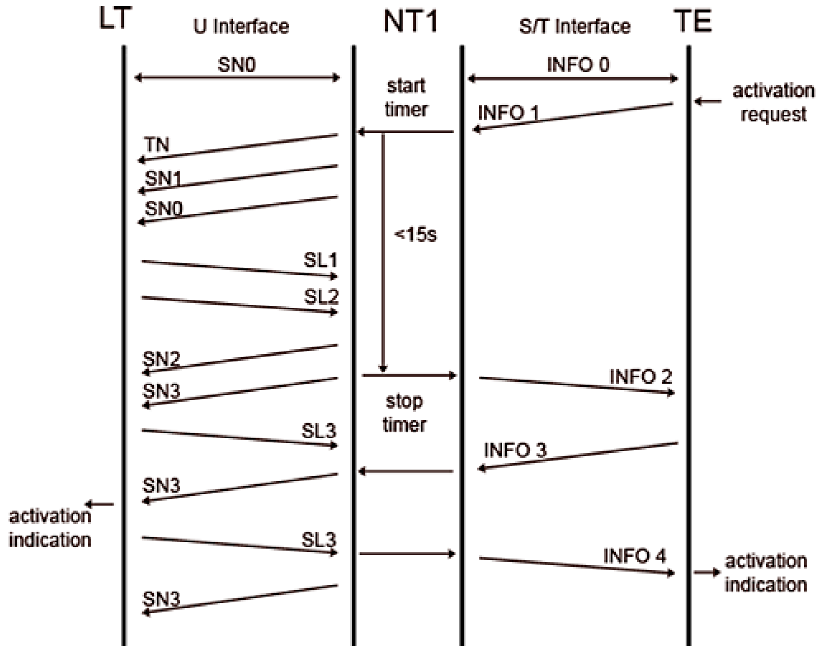

Figure 17: ETR 080 – Activation from the TE side of the 2B1Q U transmission system

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