DTMF-BASED CALLER ID (VoIP)

8.3
Most caller ID signals use FSK-based implementation. Some countries like Finland, Denmark, Iceland, the Netherlands, India, Belgium, Sweden, Brazil, Taiwan, Saudi Arabia, and Uruguay make use of DTMF-based caller ID. In practice, many DTMF-caller-ID-supported phones are also made to work with FSK-based called ID. The phone displays are capable of supporting the numbers and alphabets. In the case of DTMF caller ID, only a number display is supported.
8.3.1

DMTF Caller ID on PSTN

For the transfer of display information over analog subscriber lines, 16-DTMF digits are used in DTMF-based caller ID. Except for these special display procedures, the normal signaling procedures and physical properties for analog subscriber lines also apply to DTMF caller ID [ETSI ETS 300 659-1 (2001)]). In case of an incoming call for a subscriber, the exchange shall seize the corresponding subscriber line for the terminating call. The line seizure may be indicated to the subscriber line by means of a polarity reversal of two lines of TIP-RING or through ring alert. In line reversal, the return to the idle polarity takes place after the information transfer phase. Two different modes for the data transmission are possible with DTMF—namely data transmission before the first ring and data transmission between the first and the second ring.
Data Transmission Before the First Ringing. In this mode, the data transmission shall occur before the beginning of the first ringing pattern. Two different procedures are possible to alert the TE namely subscriber line polarity reversal or ring alert. The subscriber line polarity is not applicable with ring alert. In both cases, the first ringing has to be within the time limits of 200 to 500 ms after the data transmission is stopped. In polarity reversal, the data transmission should not start sooner than 200 ms after the polarity reversal.

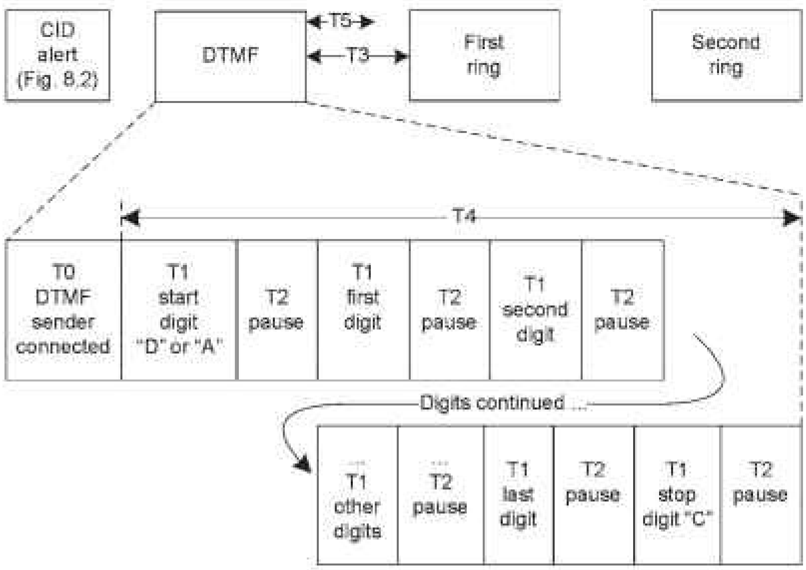

Figure 8.5. DTMF caller ID transmission.
The DTMF-based caller ID transmission method sends a series of DTMF digits before the first ringing cycle. In some implementations, alert signals are not used before data transmission in DTMF caller ID. In such cases, the central office provides a pause to connect the DTMF sender to the voice path [URL (Advent-CID2)]. As shown in Fig. 8.5, the calling number is sent after a “start” digit and ends when the “stop” digit is detected. Figure 8.5 represents the DTMF caller ID events before the

first ring. The CID alert can be either ring pulse or polarity line reversal for DTMF- based caller ID. The transfer of number information is to be regarded as complete when the DTMF stop digit "C" is received or the ringing signal is detected. The following timing details of DTMF caller ID are marked in Fig. 8.5.

TO—Pause for the central office to connect the DTMF sender to the voice path when no alert signal is preceded with DTMF transmission. TO is between 50 ms and 400 ms. When line reversal is used as an alert, the delay from the end of the alert signal to the start of the data transmission is between 200 ms and 500 ms.

T1—Duration of DTMF digit, >50ms varies by country.

T2—Inter-digit pause, >50ms varies by country.

T3—Delay from end of digits to start of first ring, 200 to 500 ms.

T4—Time required to send all DTMF digits, <3000ms.

T5—Return time to quiescent state after DTMF stop digit, <150ms.

Data Transmission Between First and Second Ring. In this mode, the data transmission shall occur during the first long silent period between two ringing patterns. The initial application of ringing is to provide an alert signal to the TE on possible data transmission. The data transmission should not start sooner than 500 ms after the first ringing pattern. At the end of caller ID data, the second ringing will start according to the normal ringing cadence. The timing of DTMF caller ID generation in the middle of the two rings will be similar to DTMF caller ID before the first ring of Fig. 8.5. In this method also, the second ringing has to be within the time limits of 200 to 500 ms after the data transmission is stopped. The caller ID number is sent after a "start" digit and ends with a "stop" digit. DTMF caller ID can display number and cannot display name.

Digits Data Coding. The DTMF transmission makes use of the 16-DTMF digit signals. Most telephones will use a keypad for the 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, # digits. The DTMF-caller-ID-capable telephone can understand incoming A, B, C, and D digits for different interpretations. The interpretation of DTMF digits by the TE during caller ID is given here.

<A> DTMF code "A" is used as a start code for the calling party number.

The start and end codes are different for different countries. Refer to country-specific PSTN recommendations for exact details on the DTMF caller ID method and timing intervals. For example, the start code is digit "A" for Brazil and "D" for Taiwan. <B> DTMF code "B" is used as a start code for the special information concerning the nonavailability or restriction information of the calling party number. It specifies the user category. <C> DTMF code "C" is used as an end code for the information transfer.

<D> DTMF code "D" is used as a start code for the diverting party number in case of call diversion.

<0 to 9> DTMF codes "0 to 9″ are used as number digits of calling/diverting party or special information code value.

<*, #> DTMF digits *, and# are not used in DTMF caller ID.

DTMF digit details and tone generation are given in topic 7. The digits and pause periods vary depending on the country. It is required to refer to Q.24 [ITU-T- Q.24 (1998)] and to local country PSTN standards for the detailed specifications of DTMF digits and pause periods.

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what-when-how

In Depth Tutorials and Information

FSK CALLER ID ON PSTN (VoIP)

8.1
The popular methods for sending Type- 1 FSK caller ID information are as follows:
1. Data transmission between first ring and second ring
2. Data transmission after alert but before actual first ring
3. Data transmission after an OSI

All these implementations use the same basic data transmission sequence with minor variations in FSK generation among ETSI and Telcordia implementations.

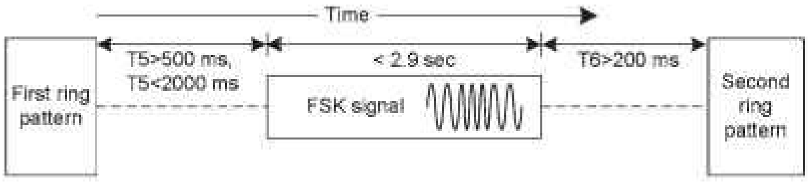

Figure 8.1. FSK Caller ID between two rings.

Case-1 Data Transmission Between First Ring and Second Ring. It is commonly used in several countries such as North America, Canada, and so on. In this signaling method, FSK modulated data are sent during the silence periods between the first and the second ring. This on-hook service provides both number and name or only name as caller ID. The ring pattern is called cadence and varies with the country requirements. Data transmission shall occur during the first long silent period between ringing patterns as shown in Fig. 8.1. The first long silent period shall be of sufficient duration for transmitting FSK data. The initial application of the ringing provides an alert signal to the TE (phone) indicating the expected data transmission. The timing for FSK and ring pattern is shown in Fig. 8.1, and this should not take more than 2.9 seconds to transmit the entire data [ETSI ETS 300 659-1 (2001)]. Usually FSK data transmitted at 1200 bps takes a very short duration compared with the available 2.9 s.

Case-2: Data Transmission After Alert but Before Actual First Ring. This type of alert-based caller ID is used in countries such as France and Japan. In this case, the data transmission occurs before the first ring. As shown in Fig. 8.2, an additional TE alerting signal (TAS) is sent before the data transmission [GR-31-CORE (2000)]. TAS signal characteristics differ between the different recommendations. In Fig. 8.2, timing intervals are marked as T, T0, T1, T2, T3, and T4, and these marking names may vary in country-specific documents. The three popular alerting signals used are dual-tone alerting signal (DT-AS), ringing pulse alerting signal (RP-AS), and line reversal followed by DT-AS
[ETSI ETS 300 659 - 1 (2001) , GR - 30 - CORE (1998)]. Japan uses line reversal followed by receiving terminal activation signal (CAR) as the alerting signal for on-hook caller ID display. The CAR signal is the receiving terminal activation signal or signal receiver seizing signal and is a short ring signal of 15-20 Hz. Refer to the NTT recommendation [URL (NTT-E)] for more details on Japan caller ID. To make caller ID work, any one of these alerting signals shall precede FSK modulation transmission within set time limits. The data transmission before the first ring using a dual-tone alerting signal is shown in Fig. 8.2(a). DT-AS shall precede FSK modulation transmission as 45 < T4 < 500 milliseconds (ms). The alerting signal DT-AS is also referred to as the CPE

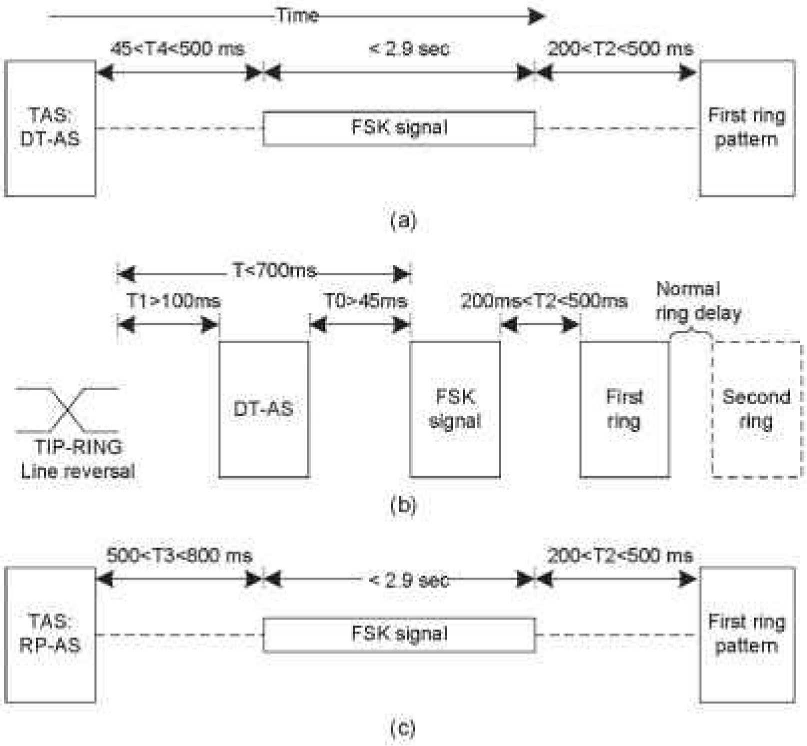

Figure 8.2. Caller ID Data Transmission Before the First Ring Using (a) DT-AS, (b) DT-AS With Line Reversal, and (c) RP-AS.

alerting signal (CAS) in some places in the document. Both names convey the same alerting signal. The data transmission before the first ring using RP-AS is shown in Fig. 8.2(c). RP-AS shall precede FSK modulation transmission as 500 < T3 < 800 ms. The line reversal followed by DT-AS shall precede FSK modulation transmission by not less than 45 ms (T0 > 45 ms) as shown in Fig. 8.2(b). The total period between the line reversal and the start of FSK modulation is less than 700 ms marked as "T." In all three methods, the application of ringing current shall start not less than 200 ms and not more than 500 ms (marked as T2) after FSK modulation transmission is stopped. The lower limits are required to enable TE to apply and remove appropriate impedance for data reception. In RP-AS, the time gap is more compared with DT-AS because of a high – voltage alert signal in RP – AS.

During caller ID, one of the three TAS signals is sent before data transmission. It varies by local PSTN standards. In this case, data transmission is associated with ringing and the FSK modulated caller ID message is followed by ring patterns. An appropriate idle condition can also be applied to the local

loop after FSK transmission. The important characteristics of DT-AS are listed here for the ETSI recommendation [ETSI ETS 300 659-1 (2001)].

Frequencies: 2130 and 2750Hz ± 0.5%

Duration: 100 ± 10 ms

Signal Level: -40dBV to -9dBV/tone

Signal purity: The total power of all extraneous signals in the voice band shall be at least 30dB lower than the power of signal fundamental frequency measured at the point of application.

Case-3: Data Transmission Alter an OSI. The OSI is the time duration when DC voltages are removed from TIP-RING lines. OSI is used as an alert to send messages and indications to the TE while the line is in the on-hook mode. This method is also known as data transmission without power ringing. In this case, the FSK modulated data are sent following a time period after OSI [GR-30-CORE (1998)]. No ringing follows the FSK data. The duration of the OSI shall be 150 to 300 ms, and the data transmission happens between 300 and 500ms after the end of the OSI. The default delay value is 500ms

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FSK CALLER ID DATA TRANSPORT PROTOCOL (VoIP)

8.2
The basic data transport protocol for caller ID is divided into four layers, namely,
1. Physical layer
2. Data link layer
3. Presentation layer
4. Application layer
The first three layers provide the actual data transport, and the application layer is used for caller ID-specific data and signaling for alerting the TE.
8.2.1

Physical Layer

The physical layer provides the interface between the caller ID service and the analog line. The physical layer provides two main functions of data transmission of service-specific information and signaling mainly for alerting the TE.
The data transmission is performed using continuous-phase FSK modulation. Data is always sent as serial binary bits in simplex mode. Data transmission is continuous, and no carrier dropouts are allowed. The start of data transmission must not corrupt the first data bit. The data transmission is
Table 8.1. Data Transmission Characteristics

Parameter	ETSI-Based Modulations	Telcordia – Based Modulations
FSK major distinction Mark (bit-1) Space (bit-0) Data rate Message format Signal generation levels Receiver operating	FSK as per ITU-T-V.23 1300Hz ± 1.5% 2100Hz ± 1.5% 1200bps ±1% Type – 1: MDMF Type – 2: MDMF -13.5 dBm± 1.5 dB, signal purity 30 dB -8 to -36 dBV (0dBV = 2.2dBm)	FSK as per Bell 202 1200 ± 12 Hz 2200 ± 22 Hz 1200 ± 12 bps Type-1: SDMF, MDMF, GDMF Type – 2: MDMF, GDMF -13.5 dBm± 1.5 dB, signal purity 30 dB -12 dBm to -36 dBm (space) -12 dBm to -32 dBm (mark)

stopped immediately after the last bit of the data-link message. The FSK data is sent asynchronously at a signal level of -13.5 dBm in both ETSI and Telcordia recommendations as listed in Table 8.1. This power level is applicable at the central office. The FSK signal level may differ for each country, because of country-specific deviations of overall loudness rating (OLR) as well as because of send and receive gain/loss planning. To get a first-level understanding on ETSI and Telcordia basic specifications, a summary is given in Table 8.1. It is suggested to refer to the ETSI [ETSI ETS 300 659-1 (2001), ETSI ETS 300 778-1 (1997)] and Telcordia [GR-30-CORE (1998)] recommendations for more details on these specifications.
8.2.2

Data Link Layer

The data link layer is responsible for providing data byte framing (start/stop asynchronous, 8-bit data) and checksum. The data link layer is also responsible for the addition of the preamble, which comprises the channel seizure signal and the mark signal as well as padding of mark bits between bytes for data flow variation.
Note that the definition of the "data link layer" varies slightly between Telcordia [GR-30-CORE (1998)] and ETSI [ETSI ETS 300 778-1 (1997)-Annex E]. The ETSI treats the message type byte and message length byte as part of the data link layer, whereas Telcordia treats the message body as a presentation layer message. Telcordia incorporates the message header (type and length bytes) as part of the presentation layer. The Telcordia recommendation refers to the presentation layer as the message assembly layer. Review the complete specifications regarding these differences.
The format of the data sequence at the data link layer is shown in Fig. 8.3(b). The data transmission between rings is presented in Fig 8.1(a) with timing information for the ETSI recommendation [ETSI ETS 300 659-1 (2001)]. The

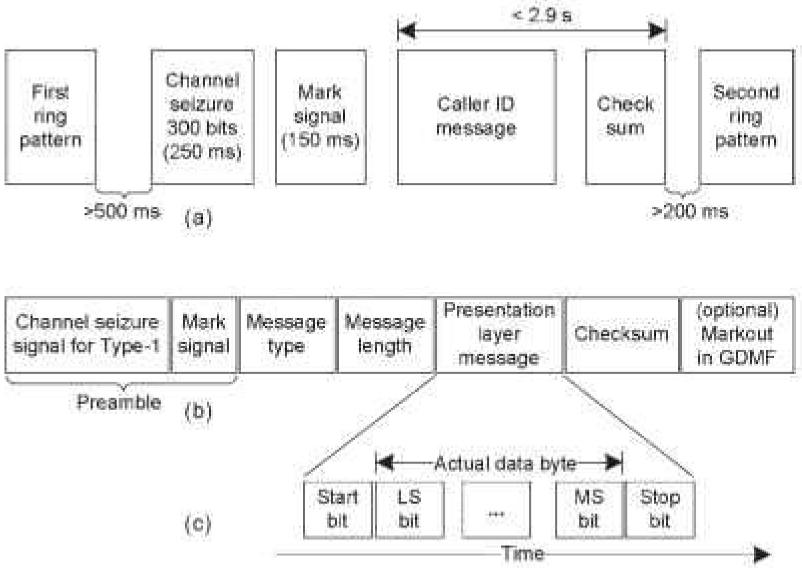

Figure 8.3. Data link layer format. (a) Details on data between two rings. (b) Message. (c) Actual data byte format.

data consist of four functional blocks—channel seizure signal, mark signal, caller ID information, and the checksum. For both ETSI [ETSI ETS 300 659-1 (2001)] and Telcordia [GR-30-CORE (1998)], every byte in the sequence (excluding the contents of the preamble, but including the checksum) is framed in an asynchronous manner with each 8-bit word preceded by a start bit (space) and followed by a stop bit (mark). The seizure signal is sent first, followed by the mark signal, and then each byte is sent in sequential order, finally followed by the checksum. No parity is used in the data framing. In the bit transmission, start bit, LS bit . . . , MS bit, and stop bit are sent as illustrated in Fig. 8.3(c). Note that actual byte is represented as stop bit, MS bit . . . LS bit and as start bits.
Channel Seizure Signal. The channel seizure signal is used to signal the start of the data transmission to the CPE or TE to alert for an incoming message, and it conditions the receiver ready for the data. This signal is used to ensure that noise is not falsely identified as a carrier. If the seizure signal is not received, the TE must cease reception, and the next messages are ignored. The seizure signal consists of 300 continuous bits (duration of 250 ms at 1200 bps) of alternating "1″ and "0." The first bit must be a "0," and the last must be a "1." The seizure signal is used only in Type-1 FSK-based on-hook caller ID and is not used in Type-2 call wait ID. The channel seizure bit size is different in ETSI and Telcordia. In some networks, the ETSI allows for a channel seizure length of 96 to 300 bits [ETSI ETS 300 778-1 (1997)].
Mark Signal. The second signal in the preamble is the mark signal. This signal consists of continuous mark bits. It is used to provide an idle period before the start of the message type byte to allow the receiver to identify the byte. For both ETSI and Telcordia, the same quantity of bits is sent as part of the mark signal. Type-1 is of 180 mark bits (duration of 150 ms at 1200 bps), and Type-2 is of 80

mark bits. Note that the ETSI allows the option of 80 mark bits for Type- 1. The tolerances on the mark signal differ between the two standards. The ETSI allows ±25 bits, and Telcordia allows ±10 bits.

Checksum. The CID information is followed by a checksum for error detection. The checksum is a single byte value, which contains a two's complement of the modulo-256 sum of all bytes in the message starting from the message type byte up to the end of the message. The channel seizure and mark signals are not included in this checksum. When the TE receives the message, it checks for errors by taking the received checksum byte and adding the modulo-256 sum of all other bytes received in the message. The checksum is verified by summing (modulo-256) all bytes in the message between the message type byte and the checksum. If the contents are correct, the checksum is zero. If the checksum fails, the entire message is discarded without any error correction or message retransmission. The TE must not send any acknowledgment or response back to the stored program control signal (SPCS). The SPCS is part of the central office.

Additional Marks (Padding). To allow for some variations in the arrival of the data to be transmitted, and to ensure that the output sequence remains continuous, a small quantity of additional marks may be added between bytes. This quantity is treated differently between the ETSI and Telcordia. The ETSI allows for additional marks between any bytes within the message [ETSI ETS 300 778-1 (1997)-Annex E], whereas Telcordia allows for insertion of additional marks between selected bytes only [GR-30-CORE (1998)]. It is an objective of the Telcordia specification that no additional marks be inserted between bytes in a parameter message. For example, mark bits (0-10) can be transmitted between message type byte and message length byte, or between parameter type and parameter length. Inserting the additional marks between parameter messages bytes is not allowed.

The number of extra mark bits (not including the stop bit) should be between 0 and 10 bits. These bits are added after the stop bit and before the start bit of the next byte. Additional marks are not permitted in the generic data message format (GDMF) payload used by Telcordia. In GDMF, to prevent corruption of the final stop bit after the checksum or last byte of GDMF, and hence destruction of the checksum, additional marks (markout) may also be used following the checksum.

8.2.3

Presentation Layer

The presentation layer is responsible for the framing of the application-specific data for provision to the data link layer. Data can be organized according to three possible formats: single data message format (SDMF), multiple data message format (MDMF), and GDMF. The ETSI recommendation supports only the MDMF format for both Type-1 and Type-2 caller ID as indicated in the message formats of Table 8.1. The SDMF is almost obsolete, and Telcordia has made it an objective that the SDMF not be used [GR-30-CORE (1998)].

The SDMF contains the calling number, date, and time. A message in SDMF includes a message type byte, a message length byte, and the actual message bytes as shown in Fig. 8.4(a). The MDMF supports multiple messages such as calling number, caller name, date and time, and other optional information. A message in MDMF also includes a message type byte, a message length byte, actual message bytes, and a parameter type and parameter length bytes as shown in Fig. 8.4(b). At certain points within these messages, up to 10 mark bits may be inserted to allow for equipment delays in the central office [GR-30-CORE (1998)]. The message type byte defines whether the message is in SDMF or MDMF. The message length byte is the binary representation of the number of bytes in the message, not including the message type, and the checksum bytes. It supports message payload lengths of up to 255 bytes. In MDMF, the parameter type byte identifies the type and format of the parameter message. The parameter length byte contains the binary representation of the number of bytes contained within the parameter message, excluding the parameter header. The maximum length of the parameter payload is 253 bytes since the maximum message length is 255 bytes, of which the parameters header consumes 2 bytes. The parameter type byte has its unique value for

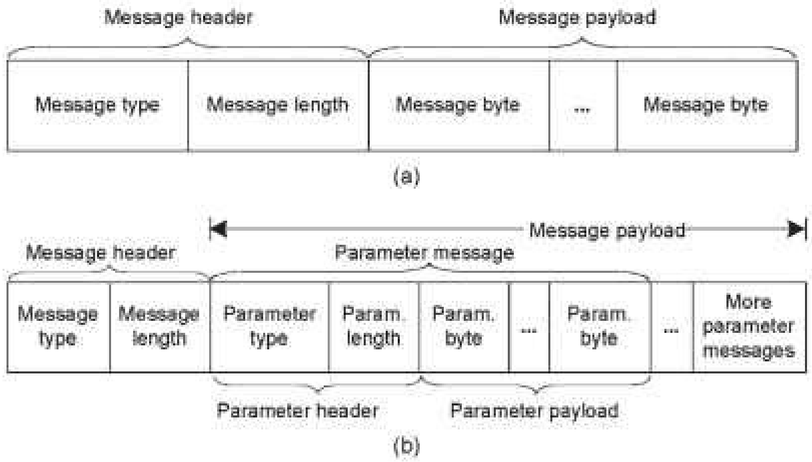

Figure 8.4. Presentation layer data formats. (a) SDMF. (b) MDMF (note: The Param. acronym is used for Parameters in the diagram).

each feature in MDMF. Refer to the ETSI or Telcordia recommendations for more details on various parameter type code words.

8.2.4

Application Layer

The application layer is where the actual caller ID service is provided. This layer determines the actual content of the data sent to the presentation layer, to suit the information to send to the TE. The application layer is country-dependent and varies between ETSI and Telcordia as well as other specification's that perform all the necessary interpretation of the data being transported, select the method caller ID display, and provide necessary timing information that is largely dependent on the specific country.

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