PERFORMANCE OF THE ISDN D-CHANNEL PROTOCOL - A SIMULATION STUDY

W. Fischer, E.-H. Goeldner
Institute of Communications Switching and Data Technics
University of Stuttgart, Federal Republic of Germany

W. Berner
SIEMENS AG, Muenchen, Federal Republic of Germany

One of the central features of the ISDN is the new user-network interface with separate channels for user and signalling information. In this paper the new standards, defined by the CCITT for the signalling channel, the D-Channel, are outlined. The capacity of 16 kbit/s allows to share the channel between signalling and low speed packet switched data. This bears a heavy impact to the prioritized signalling. Furtheron, up to 8 Terminal Equipments (TE) in varying configurations require new protocol elements with unknown performance behaviour. The implementation of the subscriber interface is shown by the Siemens EWSD system. Out of this, a detailed simulation model has been developed, comprising all features of the D-Channel protocol and of the implementation in the real system. The simulation results focus on the call set-up only and show the impact of an increasing load by packetized data to the call delay and to the processor loads at the exchange termination. As a parameter, the number of TEs varies from 1 to 8 in the presented results.

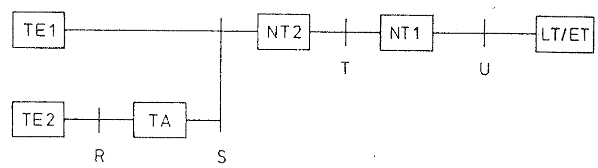
1. INTRODUCTION

Within the recent study period (1981-1984), the CCITT study groups elaborated the framework of recommendations for the international use of integrated services digital networks (ISDN). The ISDN, as the consequent evolution of digital transmission and switching, comprises circuit switched services like normal voice transmission, text and data services based on 64 kbit/s channels, broadband services as well as packet switching capabilities. One of the main features of ISDN is the provision of full digital end-to-end connectivity. Only this allows the variety of new services without restriction for further requirements. Due to this, a user-network interface has been defined allowing the use of all network capabilities. At the users' interface, only one ISDN is visible although inside an ISDN, several separate networks will realize the different network capabilities (e.g. one network for 64 kbit/s circuit switching, another for broadband, and a third for packet switching). Therefore, the integration of the different services must be accomplished on the subscriber loop.

Transmission of signalling information must be possible at any time, even simultaneously with a call and without any influence on user data. These goals are reached by a separate signalling channel, the D-channel, with 16 or 64 kbit/s. The user-network-interface has been defined by a reference configuration (Fig. 1)

In the following, we shall focus on the "Basic Access" only, the access of an ISDN subscriber by a single loop, providing two B-channels (64 kbit/s each) and one D-channel with 16 kbit/s.

The subscriber loop is denoted by reference point U. The reference point S, a short passive bus, will be the interface for the different terminals, allowing up to 8 terminals to be connected. The use of the two B-channels is controlled by signalling on the D-channel.



TE1: Terminal Equipment with ISDN-Interface
TE2: Terminal Equipment with any other Interface (e.g. X.21)
TA: Terminal Adapter
NT1: Network Termination for Functions of Layer 1
NT2: Network Termination for Functions of Layers 2 and 3
LT: Line Termination for Functions of Layer 1
ET: Exchange Termination for Functions of Layers 2 and 3
R, S, T, U: Reference Points

Fig. 1: User-Network-Interface
Reference Points and Functional Entities

Signalling is done by frame-oriented messages, defined by a layered protocol model. The bandwidth of the D-channel, not required by signalling messages, can be used for low-rate packet switched data, according to Rec. X.25 [4] or for teleaction information.

2. THE D-CHANNEL PROTOCOL

Layer 1 of the D-channel protocol [1] specifies the interface at the S-bus. The access of the different TEs to the D-channel is controlled by a kind of carrier sense multiple access protocol with collision detection (CSMA/CD). All information sent by the TEs on the D-channel is echoed at the NT on a separate D-echo-channel. This scheme allows the stations to detect a collision immediately. The station sending the first '0'-bit claims priority against all others and keeps on sending without any loss of data; all other stations withdraw their messages.

Layer 2 of the D-channel-protocol is based on the LAPB-procedure of X.25 (HDLC-ABM) [4], adopted and extended to the new requirements of this point-to-multipoint configuration [2]. The frame

Contribution to the 8th ICCC
Munich, F.R.G. September 15 - 19, 1986

format corresponds to the HDLC-frame format; only the address field has been extended to comprise the Service Access Point Identifier (SAPI, 6 bit) and the Terminal Endpoint Identifier (TEI, 7 bit). The TEI allows several signalling links to be established in parallel at a time on one D-channel. The SAPI is needed to indicate the type of information a frame is containing: signalling information, packet switched data, teleaction information or management information for the assignment of TEI values to the active TEs.

Plugging in or out a TE is possible at any time, and therefore, the local exchange does not know the present terminal configuration at the subscriber line. This led to the following definition of the protocol for setting up a signalling link: only a TE can set up a logical link, using the HDLC-frame 'SABM'. An incoming call is indicated by the exchange using a broadcast message to all active TEs connected to this S-interface. All TEs, being able to accept the call set up a signalling link and answer the call. The first TE accepting the call will be connected via the B-channel. All other TEs are cleared, and their signalling link is released.

The broadcasted 'incoming call' is transmitted using unnumbered information (UI) frames (Unacknowledged Operation Mode). On an established link frames are flow-controlled by a sliding window mechanism according to the rules of HDLC (Multiple Frame Acknowledged Operation Mode). For LAPs transmitting signalling information, the window size will be limited by 1.

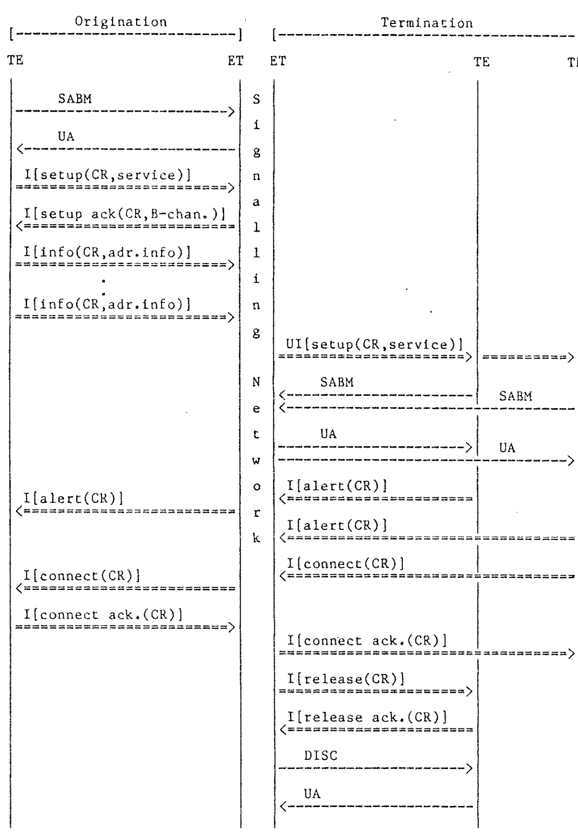


Fig. 2: Signalling Scheme for Call Set Up (CR: Call Reference Number)

Layer 3 defines the signalling procedures between the TE and the ET for controlling connections an a B-channel and for the use of service facilities [3]. An example for the various signalling schemes is shown by Fig. 2.

The originating TE first sets up a signalling link, using the unnumbered frame 'SABM', acknowledged by the frame 'UA'. After successful establishment of the link signalling messages (Layer 3) are then transported within I-frames. To identify the adjoint call, every signalling message includes an arbitrarily chosen 'Call Reference Number' (CR). By a 'Setup' message, the desired service is indicated. The 'Setup Acknowledge' message assigns a B-channel, and with the following 'Info' messages all information needed for setting up that call are sent.

At the terminating TE, the incoming call is indicated by an unnumbered information frame (UI-Frame) with group address. All TEs, able to accept this call have to set up a logical link first (SABM - UA). As in our example for a telephone call, all TEs are signalling back that they are 'Alerting'. That TE accepting the call indicates this by sending a 'Connect' message. The exchange will acknowledge this to both TEs - originating TE and terminating TE. The other terminating TEs which did not get that call will be released and stop alerting. The signalling link to this TE is now cleared down using the 'DISC'-frame. The logical links to and from the TEs which are involved in this call are kept active until the B-channel connection is released.

3. THE IMPLEMENTATION IN THE SIEMENS EWSD SWITCHING SYSTEM

3.1 Overview

The EWSD-System has been extensively described in the literature (e.g. [6,9,10]). An overview will, however, be given of these elements which are directly affected by the introduction of ISDN. A simplified EWSD block diagram is shown in Fig. 3.

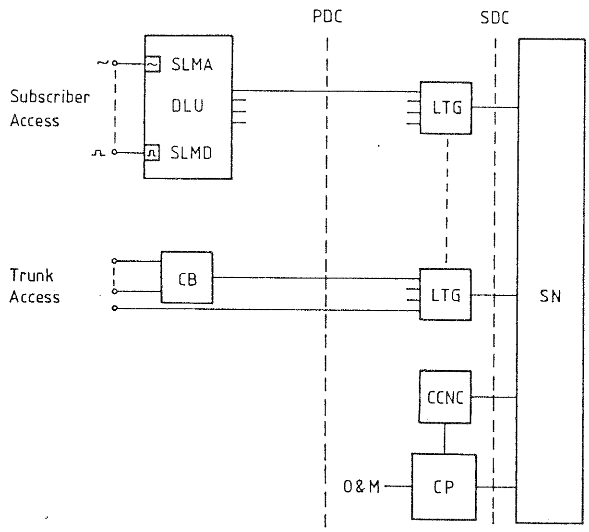
EWSD is a fully digital switching system, using a distributed control architecture, which facilitates a broad application range as well as the introduction of ISDN. Internal message handling capabilities of distributed switching systems like EWSD are very important for its call handling capacity especially regarding the requirements of ISDN. Most of the signalling and call processing functions (Layer 3 of D-channel protocol) of the EWSD are performed by the group processors in the Line/Trunk Groups (LTG). These group processors provide the control for the signalling from the outside world but internally offer a uniform logical interface to the switch. The Coordination Processor (CP) primarily performs Operational and Maintenance (O&M) functions and call coordination tasks using the well-defined logical interface to the LTGs which protects the CP from the direct impact of the changing environment.

The Digital Line Unit (DLU) provides the interface to the subscriber loops analog or digital:

* Analog subscriber lines are connected by the Subscriber Line Module Analog (SLMA).

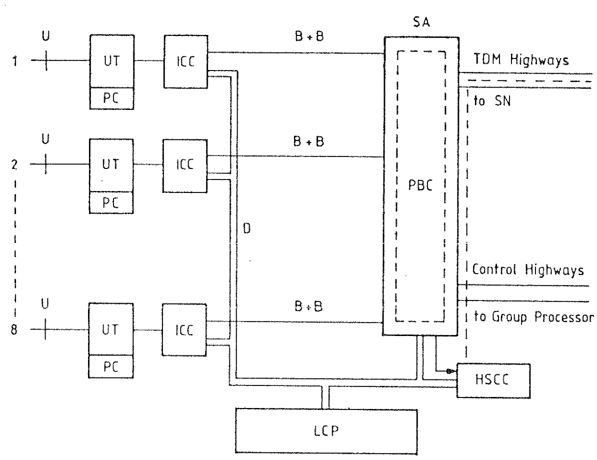
* The basic access is connected by the Subscriber Line Module Digital (SLMD).

The basic function of the DLU is to concentrate subscriber traffic and to offer a uniform interface to the rest of the system via one or more Primary Digital Carriers (PDC). This permits it to be located remotely from the switch or locally. A subset of CCITT Signalling System No. 7 (SS7) is used between the DLUC (the controller of the DLU) and the EWSD system for both signalling and control purposes.



CB: Channel Bank
CCNC: Common Channel Network Controller
CP: Coordination Processor
DLU: Digital Line Unit
LTG: Line/Trunk Group
O&M: Operation and Maintenance Center
PDC: Primary Digital Carrier
SDC: Secondary Digital Carrier
SLMA: Subscriber Line Module Analog
SLMD: Subscriber Line Module Digital
SN: Switching Network

Fig. 3: EWSD Block Diagram



HSCC: High Speed Communication Controller
ICC: ISDN Communication Controller
LCP: Line Card Processor
PBC: Peripheral Board Controller
PC: Power Control
SA: System Adapter
SN: Switching Network
UT: U-Transceiver

Fig. 4: Block Diagram of the SLMD

3.2 The Digital Line Card SLMD

While layer 3 processing of the D-channel protocol is provided by the group processor, the layers 1 and 2 are handled on the peripheral line card SLMD (s. block diagram Fig. 4, for a more detailed description see [8]). Thanks to the VLSI-devices 8 basic accesses can be accommodated on this module. As shown in the block diagram the UT-devices (layer 1), power feeding circuits and the ICC, separating the B-channel and supporting the basic layer 2 protocol functions, are provided on a per-line basis.

The B-channels are routed to the Switching Network via the Peripheral Board Controller (PBC), while the D-channel is routed to the Line Card Processor (LCP) for handling the layer 2 protocol and separating s- and p-data.

The p-data are forwarded via the HSCC to a dedicated 64 kbit/s time slot. For this purpose a nailed connection between HSCC and a Service Module (SM) for p-data handling is provided by the Switching Network (SN). At least one 64 kbit/s time slot per DLU has to be assigned for p-data. The signalling information from the D-channel is assembled by the LCP and communicated to the DLUC via an internal DLU protocol over a high speed bus.

The ICC chip has two 32 byte FIFOs for each transmission direction. This allows to store two times 32 bytes of D-channel information or 2 complete HDLC-frames with up to 32 bytes. Most of the processing power of the LCP - a 16 bit µP - is needed for performing the layer 2 protocol. Futhermore, the realtime condition for serving the FIFOs of the ICC must be taken into account.

4. THE SIMULATION MODEL

A first step to get an understanding of the total system in terms of processes, their mapping onto processor phases and the inter-process communication has been the definition of the protocol structure.

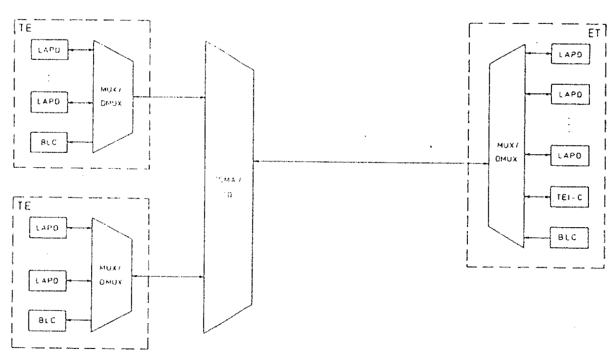


Fig. 5: Protocol Structure

On the left hand side of Fig. 5 there is a number of TEs each of them containing in general a number of LAPs. In each TE these LAPs get their access to the S-interface by means of a MUX/DEMUX procedure.

The transmission of unnumbered information frames is controlled by means of the BLC (broadcast link control) process in the TEs as well as in the ET.

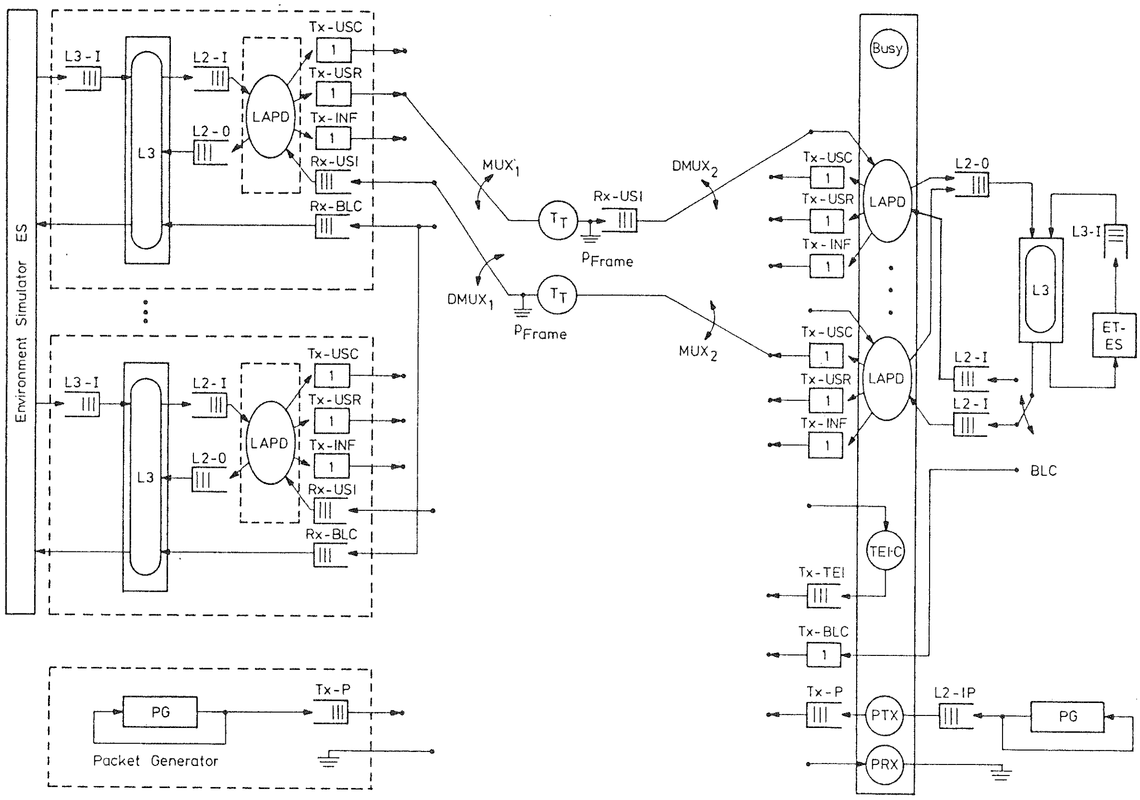


Fig. 6: Simulation Model

The channel access procedure, a CSMA/CD protocol, has also been depicted like a multiplexer as there are two mechanisms which guarantee that exactly one frame will be transmitted via the S-interface without delay. The first mechanism is the priority scheme which ensures the higher priority of signalling information compared to packetized data and the fair scheduling of all connected terminals. The second one guarantees in the case of a collision that this terminal sending the first "zero-bit" will succeed and transmit its frame while the other one(s) will withdraw and try again later.

On the right hand side of Fig. 5, in the ET there are as many LAPs as in all connected TEs together to ensure point-to-point connections between pairs of LAPs. The TEI-C process controls the assignment or confirmation of TEIs. The access of all these processes to the U-interface is by means of a MUX/DEMUX-procedure. The TEI management in the TEs is a matter of the LAPs.

From this protocol structure we derived the simulation model as depicted in Fig. 6.

On the left hand side of Fig. 6 there is again a number of TEs each containing now only one LAP-process for signalling information as it will be the case in standard applications in the ISDN and because we combined both multiplex mechanisms mentioned above to one procedure ($MUX_1/DEMUX_1$) and it can be shown that this procedure will handle all signalling LAPs identically, no matter how many LAPs per TE are there.

Buffer	Type of Frame*		Overwrite	Priority
Tx-USC	U/S	Command	Yes	1
Tx-USR	U/S	Response	Yes	2
Tx-INF	I	Command	No	3

(* according to LAPB-specifications)

Table 1

Each LAP has one input buffer and three output buffers connecting it to the $MUX_1/DEMUX_1$, each of the latter three with one place. Their function shows Table 1. The buffers of one LAP are scheduled according to the priority given in Table 1, while the LAPs as a whole are scheduled according to a cyclic polling strategy.

The packet generator which generates a basic load of packetized data will be scheduled with lowest priority.

The LAPs in the TEs are connected to the layer 3 process (L3) by a pair of buffers (L2-I, L2-0).

L3 is running on a separate processor and is controlled by the L3 protocol. The buffers Rx-BLC (BLC = Broadcast Link Control) are used for the reception of UI-frames with group address (e.g. UI[setup]). The BLC processes have been included into L3.

The behaviour of the user of each TE is controlled by an environment simulator ES. Its behaviour is characterized by an alternation of service- and idle times.

On the right hand side of Fig. 6 in the ET there are two processors, one for the L3 process which is controlled by the L3 protocol, and one for the L2 processes. These processes are

* LAP (as many as there are TEs connected)
* PTX, the sending phase for packetized data
* PRX, the receiving phase for packetized data
* TEI-C, the TEI management process
* Busy, the phase representing the other user-network-interfaces handled by the same L2 processor. The busy phase is scheduled after each L2 phase and its duration is a function of the previous phase.

The buffers connecting the L2 processor with the U-interface are scheduled in the same way as described for the TEs. The priorities of LAPs, TEI-C, BLC and PTX are arranged in descending order.

The channel is mapped only by a phase representing the transmission time of a frame T_T and the probability of a frame to get lost because of a transmission error p_{Frame}. The propagation delay has been neglected because for the given problem it is some orders of magnitude less than the transmission delay. The times for establishment and release of the layer 1 connection have been neglected too as they represent only an additive constant in the call delay.

A typical scenario for a connection establishment is shown by Fig. 2. Considering this scenario and the frequency of call attempts on a single D-channel it can be assumed that in the average of time the load by signalling information can nearly be neglected.

But the scenario also shows that each incoming setup message can cause a set of terminals having the same service features to respond by setting up a link and signalling "alerting". While this happens, the channel and perhaps the processors too, can represent a bottle-neck which increases considerably the call delay. An additional very strong influence will be given by a basic load of packetized data.

5. SIMULATION RESULTS

In advance of the discussion of the results a few comments are necessary dealing with the input parameters. In order to obtain a high number of relevant events within reasonable simulation time, we assumed some time values (e.g. the transmission time via the signalling network) lower than they would be in real applications. This has to be taken into account when the results are interpreted.

As we did only simulate the behaviour of one basic access we had to distinguish between the two directions of connection establishment (TE->ET, ET->TE).

The main results which are of interest in the scope of this paper are

* the call delay which is the time between
    - Sending of the last I[info] and receiving of I[alert] by the originating TE in the case of a calling TE (we assumed the time composed of 2 * (transmission delay through the signalling network) + (call delay in the case of a calling ET) to be only 50 ms).
    - Sending of UI[setup] and receiving of the first I[alert] by the terminating ET in the case of a calling ET.
* The dial tone delay, which is the time between sending of SABM and receiving I[setup] by the calling TE.
* The occupancy of the ET-L2 processor and the portion of the diverse phases.

All the figures show the results versus the basic channel load by packetized data. The meaning of the parameters of each curve is as follows:

* ET -> xTE: direction of connection establishment ET -> TE, number of x TEs responding
* TE -> ET: direction of connection establishment TE -> ET

The 95% confidence intervals are so small that it has not been worth including them in the figures.

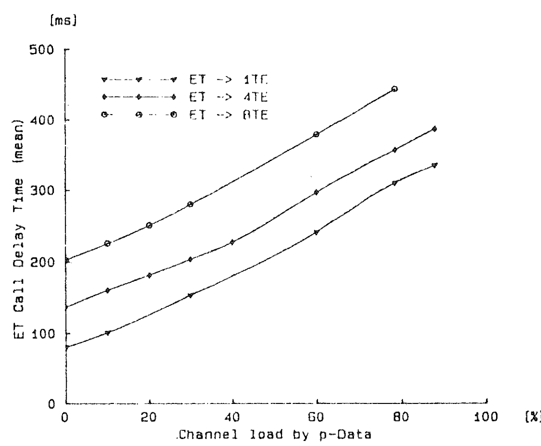


Fig. 7: Mean Call Delay (ET calling)

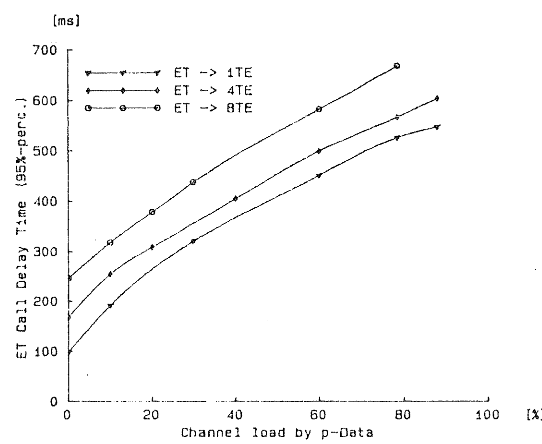


Fig. 8: 95% Percentile of Call Delay (ET calling)

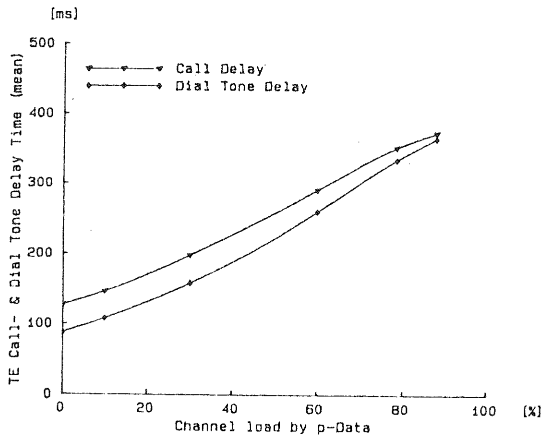


Fig. 9: Mean Call- and Dial Tone Delay (TE calling)

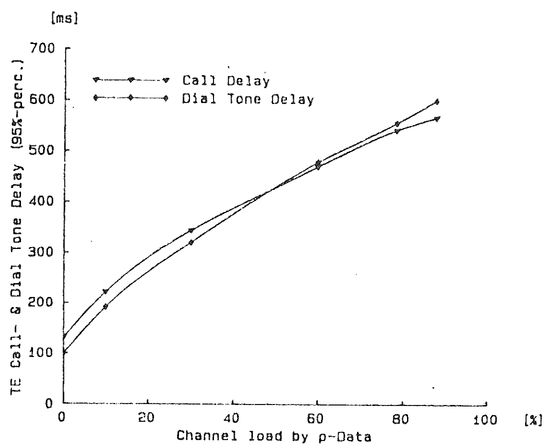


Fig. 10: 95% Percentile of Call- and Dial Tone Delay (TE calling)

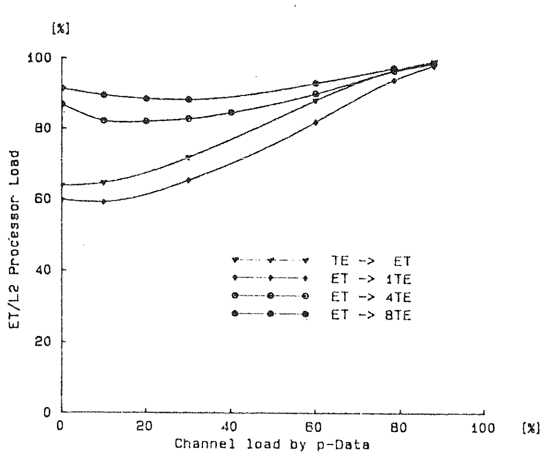


Fig. 11: ET/L2 Processor Load Measured Within Call Delay

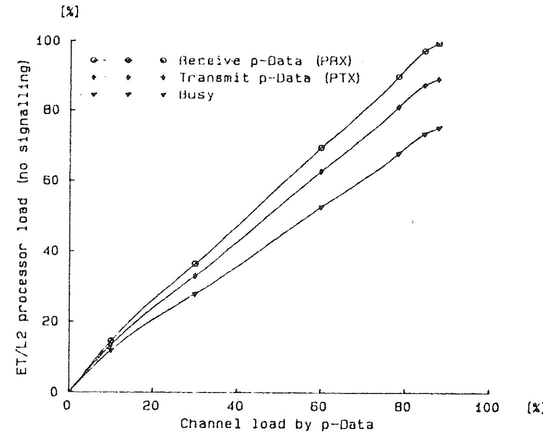


Fig. 12: ET/L2 Processor Load (Only p-Data) Portion of the Diverse Phases

Table 2 shows the input parameters which we assumed for the simulation.

Parameter	DF*	Mean Value
Holding times		
LAP (ET) TE calling	E_8	4.0 ms
ET calling	E_5	3.6 ms
LAP (TE) TE calling	E_9	3.5 ms
ET calling	E_8	2.6 ms
PRX (ET)	E_{12}	5.6 ms
PTX (ET)	E_{13}	8.4 ms
L3 (ET/TE)	D	3.0 ms
T_T	D	Frame length / 16kbps
Busy	3 * prev. phase + 0.5 ms	
Packetized Data		
packet length	M	800 bits
interarrival time	M	variable
Transmission errors		
P_{Frame}	Resulting from independent bit errors of $p_{Bit} = 10^{-5}$	
Miscellaneous		
Length of I-frames	D	according to [5]

DF*: Distribution Function

Table 2

The values for the holding times have been derived from a test implementation. The function of the busy phase is a worst case consideration for a number of 4 U-interfaces served by a single L2-processor.

To get the overall call delay take call delay for TE -> ET, subtract 50 ms, add 2 * estimated transmission delay via signalling network, add call delay for ET -> xTE.

6. CONCLUSION

We have presented a simulation study of the D-channel protocol which is based on an implementation in the Siemens switching system EWSD for the ISDN pilot project of the Deutsche Bundespost. We obtained results for the call delay and the processor occupancies for diverse configurations under the assumption of additional packetized data traffic on the D-channel which will not yet be part of the pilot project but will certainly be implemented in the near future.

The results presented are obtained under worst-case assumptions. If we assume normal operating conditions (i.e. not all users are active simultaneously), processor occupancy and reaction times will decrease, but even under the assumptions considered, it can be said that the D-Channel protocol will perform well if the channel load by packetized data will not exceed a limit which is due to the maximum values for delay times specified by CCITT.

7. ACKNOWLEDGEMENTS

We wish to thank R. Stiefel and W. Schollenberger for their effort in the development and implementation of the simulator and W. Dieterle for the fruitful discussions in the initial phase of the project.

8. REFERENCES

[1] CCITT Recommendations I.430, I.431 (layer 1 spec.); Red Book, Geneva, 1984

[2] CCITT Recommendations I.440, I.441 (layer 2 spec.); Red Book, Geneva, 1984

[3] CCITT Recommendations I.450, I.451 (layer 3 spec.); Red Book, Geneva, 1984

[4] CCITT Recommendation X.25; Red Book, Geneva, 1984

[5] National Recommendation 1R6 of the Deutsche Bundespost, Darmstadt, 1984

[6] Botsch,D.: Enhanced Features and Applications for EWSD, ISS-81 Montreal p.23A4-1, Telcom Report, Vol. 4 (1981)

[7] Koenig,W.; Truong,H.L.; Waldmann,G.: D-Kanal-Protokoll im ISDN Pilotprojekt, NTG-Fachberichte 88, VDE-Verlag Berlin 1985

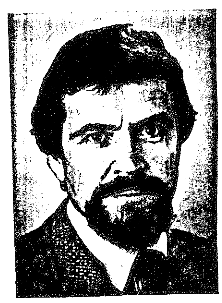
[8] Lutz,K.-A.; Berner,W.; Tannhaeuser,A.: ISDN-Leitungsanschlusseinheit, NTG-Fachberichte 88, VDE-Verlag Berlin 1985

[9] Skaperda,N.; Ribbeck,G.: EWSD as a Basis for ISDN, ISS-84 Florence p.21B3-1

[10] Suckfuell,H.: Architecture of a new line of digital switches, ISS-79 Paris p.221

Wolfgang Fischer received his Dipl.-Ing. degree in electrical engineering in 1983 from the University of Stuttgart, F.R.G. Since this time he has been working as an assistant professor at the Institute of Communications Switching and Data Technics, University of Stuttgart, where he is involved in performance investigations of the ISDN, switching network analysis and in formal specification techniques for communications protocols.

Ernst-Heinrich Goeldner received the Dipl.-Ing. degree in electrical engineering in 1981 from the University of Stuttgart, F.R.G. Since 1982 he has been working as an assistant professor at the Institute of Communications Switching and Data Technics at the University of Stuttgart. His interests include the field of integrated inhouse communication, performance of communication protocols, ISDN and LAN aspects.



Wolfgang Berner received his Dipl.-Ing. degree in electrical engineering in 1973 from the University of Stuttgart, F.R.G. Since 1974 he has been with the Siemens company where he was involved in the development of the data switching system EDS until 1979. Since then he has been engaged in the design and development of new communications systems, e.g. definition and implementation of protocols for the ISDN user-network interface. His present main interest is the development of the broadband ISDN.