

**ARCHITECTURE OF A NEW INHOUSE COMMUNICATION SYSTEM
PROVIDING INTEGRATED CIRCUIT AND PACKET SWITCHING**

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ABSTRACT

A new inhouse communication system is suggested, integrating circuit switched services with variable bandwidth and packet switched services with variable throughput rate. The concept is based on the existing subscriber lines for ISDN services, and extended by small LANs and new PBXes, which are interconnected through broad-band links. LANs, PBXes and broad-band links are appropriate for integrated circuit switching and packet switching. In the first part of this paper, integration principles within the PBX as well as implementation aspects are outlined. The second part compares performance results of several strategies to achieve the integration of both switching principles on a ring system and presents a pilot implementation.

1 INTRODUCTION

Today's inhouse communication is characterized by two mainstream developments, shown in Figure 1:

- Pure packet switched (PS) local area networks (LAN) for computer communications, and
- Full-digital private branch exchange (PBX) systems for circuit switched (CS) traffic, e.g. PCM coded voice.

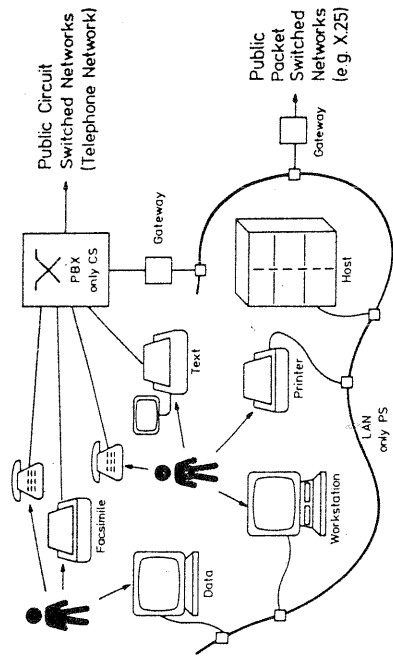


Figure 1: Today's Inhouse Communication

Future PBXes will be extended to include circuit and packet switched text and data traffic, matching the specifications given by the Integrated Services Digital Network (ISDN) with transmission over one or two 64 kbps channels (B-Channel) and a separate 16 kbps channel (D-Channel) for signalling and low-speed PS-data using existing subscriber lines. Arbitrary connections between terminals or end systems of different networks, as well as the connection of the private network to public networks require special gateways.

This technique maintains the characteristic features for each class of application: CS for stream type communications like voice or mass data transfer and PS for interactive (bursty) data communication. But new developments within the area of terminal equipment, intelligent workstations, office and factory automation require a universal communication network integrating CS-services with variable bandwidth and PS-services with high throughput rate.

Today neither a LAN, nor a PBX can satisfy these new requirements, because

- Normal LAN's are often not suitable for stream type traffic (e.g. voice, video) due to the load dependent waiting times.
- Normally, PBXes have only narrow-band subscriber lines which give a limitation of the maximum bandwidth, leading to large delays for bursty traffic with high-speed transmission rates. Additionally, compute and file servers maintain often many connections simultaneously which cannot be realized by CS-PBXes satisfactorily.

Different networks will be interconnected by Gateways with large buffer requirements and high protocol conversion overhead, resulting in a throughput bottleneck. Therefore, it is necessary to develop a new network concept without all these drawbacks.

2 NETWORK CONCEPT

We assume that even in future the mass of terminals will be of the ISDN-type for voice, text and data, which will be connected through existing subscriber lines to centralized ISDN-PBXes. But in smaller spacial areas, such as a department for research or development, or an university institute there will be higher communication requirements through the simultaneous use of voice, data and graphic communication, caused by multi-functional workstations, department computers, data bases and graphic equipments.

To satisfy all these new requirements we have developed a new inhouse communication system which provides

- synchronous transmission for CS-traffic (e.g. voice, video and mass data transfer) with variable and adaptable bandwidth as well as
- asynchronous transmission of data packets (e.g. dialog-oriented applications) with high throughput rate.

Figure 2 shows the basic structure of the new system which deals with the following aspects:

- Small LANs (SLAN) on a ring basis for the integration of CS-and PS-traffic installed in areas with higher communication requirements. Fixed time slot allocation is provided for synchronous transmission of voice and video whereas for data synchronous (e.g. file transfer) as well as asynchronous (interactive data transfer) transmission can be selected. Higher bandwidth for CS will be

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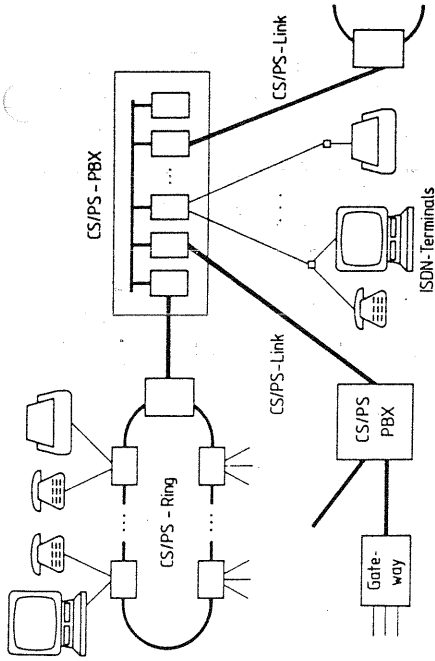


Figure 2: Basic Structure of the New Inhouse Communication System

achieved through the allocation of several time slots whereas the bandwidth, not used by CS is available for PS and allows a high throughput rate for PS-traffic.

- Several SLANs (e.g. within a plant) will be interconnected through a new type of PBXes integrating CS and PS with the same features as mentioned above. Inside larger inhouse areas several CS/PS-PBXes may be interconnected in a mesh-type structure through integrated CS/PS-Links.
- Even in future most terminals will be of the ISDN-type. They will also be connected to the new CS/PS-PBXes in the usual star configuration through existing subscriber lines using ISDN interfaces.

The advantages of this new inhouse communication system are:

- Maintaining of the adequate service-specific switching principles (CS or PS).
- Limitation of distributed functions to very small areas (SLAN) where needed.
- Handling of mass traffic by centralized switching nodes.
- Matching of specific grade of service criteria as throughput and delay for PS-services and blocking for CS-services.
- Concentration of all traffic to public networks to one gateway.
- Embedding of the new structure into the existing infrastructure.
- Distributed and centralized system management functions.

3 INTEGRATED CS/PS-PBX

3.1 Basic Requirements

The inhouse communication system presented in this paper consists of small CS/PS-LANs, CS/PS-PBXes and CS/PS-Links for interconnecting the elements. The CS/PS-PBX has to satisfy all the following requirements:

- The node must be suitable for CS with variable bandwidth as well as for PS with high throughput rate.
- Interconnection of asynchronous end systems with different bit rates and synchronous end systems with variable bandwidth must be possible.
- Several integrated CS/PS-SLANs may be connected to a CS/PS-PBX.
- In larger inhouse areas more than one CS/PS-PBX will be installed and will be interconnected through CS/PS-Links in a mesh-type structure.
- Connection to foreign exchanges or public networks by gateways must be provided by the CS/PS-PBX.

3.2 Principles of Integration in the PBX

CS-services and PS-services can be integrated in the switching node by different principles which will be addressed below.

3.2.1 CS/PS-Integration by a Circuit Switching Node

The PBX can be realized as a CS-node shown in Figure 3. PS-services are provided by datagrams or virtual connections, both requiring terminal adaptors (TA).

Using datagrams, first a CS-connection will be set up and after transmitting the packet, the CS-connection will be cleared down. For virtual connections the packets may be switched in the same manner as datagrams (Fast Circuit Switching [2], Burst Switching [3]), or every virtual connection will be mapped to a physical (CS) connection. Both modes are not well suited for PS because setting up and clearing down a CS-connection per packet results in a large overhead, whereas a CS-connection per virtual connection limits the total number of possible connections and therefore the total throughput may not be very high. Every virtual connection requires a CS-connection and no multiplexing is possible on one line.

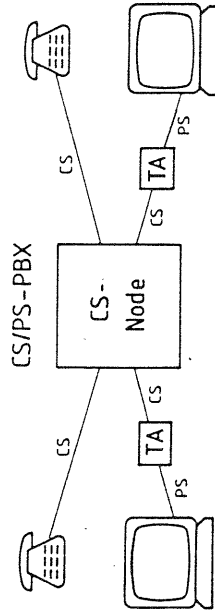


Figure 3: CS/PS-Integration by a Circuit Switching Node

3.2.2 CS/PS-Integration by a Packet Switching Node

CS/PS-integration in the PBX may also be realized by a PS-node [4] as shown in Figure 4. The synchronous data must be assembled to a packet by the transmitter and after transferring the packet through the network it has to be disassembled by the receiver. Not all realtime applications (synchronous transmission) are suitable for PS due to the load dependent waiting times.

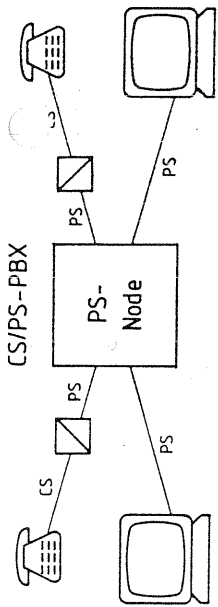


Figure 4: CS/PS-Integration by a Packet Switching Node

Voice transmission can be done by PS because small variable delays and even occasional loss of packets will not influence the voice communication drastically. Today's switching nodes are not suitable for packetized voice because their capacity is not large enough and not all used protocols are necessary for packetized voice. But the development in transmission and switching techniques leads to the Fast Packet Switching [2] principle with simpler protocol features and more powerful switching nodes.

3.2.3 CS/PS-Integration by two Switching Nodes

Both principles discussed above use only one switching mode, CS or PS, resulting in additional hardware and software overhead for terminal adaptors or CS/PS-transformation units and therefore the characteristic features of PS-services and CS-services cannot be maintained, respectively. The following principles eliminate these drawbacks.

A simple kind of integration in the PBX is the use of a CS-node and a PS-node located together as shown in Figure 5. An arbitrary connection of different terminals or end systems will be possible by bridging the CS-node and the PS-node through gateways. The advantage of this principle is that the specific features of the CS-services and those of the PS-services will be maintained but gateways normally result in a throughput bottleneck, and only part of the node functions may be shared by each of the service types.

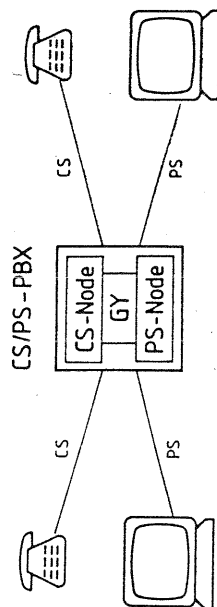


Figure 5: CS/PS-Integration by two Switching Nodes

3.2.4 CS/PS-Integration by a Hybrid Switching Node

Hybrid switching (HS) combines both CS and PS. It maintains the characteristic features of both services and needs no PBX-internal gateway (Figure 6). The integration can be achieved by using one internal transmission medium for all kinds of traffic [5]. HS is based on a synchronous pulse frame with fixed length. This frame is partitioned into equal-sized time slots similar to the well-known PCM-frame. One time slot is able to carry one CS-channel (64 kbps). Time slots to a new CS-call

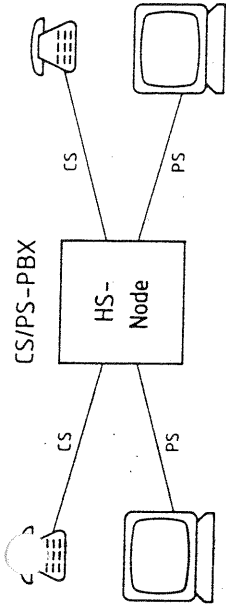


Figure 6: CS/PS-Integration by a Hybrid Switching Node

are allocated at call set up through a signalling procedure. The different integration possibilities within the switching node are similar to the integrated LAN and have been published in [1].

3.3 CS/PS-PBX Implementation (Bus)

The CS/PS-PBX shown in Figure 7 will be realized as a synchronous, bitparallel bus system (data-, address-, control- and packet access bus) with several access modules (AM) and one master module (MM). Subscriber lines or CS/PS-Links are connected to the CS/PS-PBX through interface modules within the AMs. A similar node structure, but with a different operation principle, has been suggested in [6].

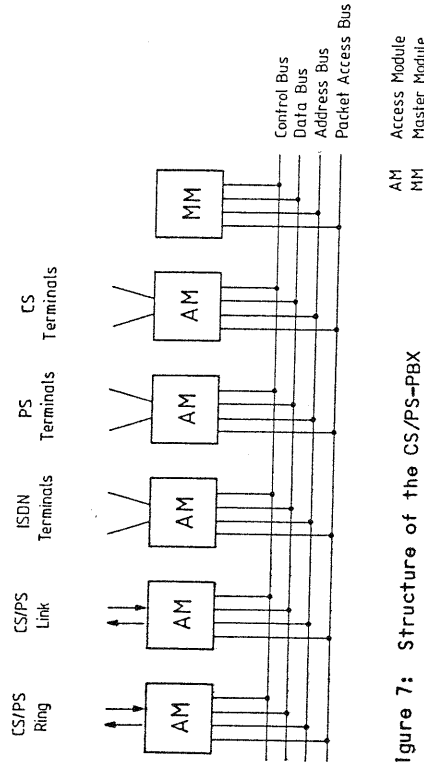


Figure 7: Structure of the CS/PS-PBX

The synchronous pulse frame used within this system is partitioned into equal-sized time slots (Figure 8). One of the time slots is reserved for broadcast messages or maintenance, whereas all other time slots can be used by CS-traffic or PS-traffic. One time slot is able to carry one simplex CS-channel with 64 kbps transmission rate. All these time slots not being used by CS-calls (empty time slots) can be considered as being concatenated to one remaining PS-channel. The access to this PS-channel is controlled by the Packet Access Protocol implemented on a separate bus system, the Packet Access Bus.

3.3.1 The Master Module

The MM is responsible for initializing all AMs as well as for generating the system clock and the pulse frame structure. For CS the MM performs connection set up

4.1.3 Minipackets

Our new integration principle discussed here uses the same strategy for CS-traffic as ISC. But the remaining bandwidth for PS-traffic is not given to one station as in case of the token passing protocol. Messages delivered from a terminal connected to a ring station will first be segmented into equal-sized and individually addressed minipackets (MP). Each minipacket fits exactly into one time slot. Every time slot is marked by two bits, indicating whether this time slot is available for PS and whether this PS slot is empty.

In most protocols PS-messages are sent around the ring, copied by the receiver and removed from the ring by the sender. In contrary to these protocols, empty slots are now used on demand by the sending station, whereas the receiver is responsible for the clearing of the slot. Therefore, a station being prepared to send PS-data within minipackets, watches the slots passing by and inserts the minipackets into unused slots. A station detecting its own address in the header of a minipacket copies the contents into its buffer, sets the time slot empty immediately or may even insert an own minipacket to be sent. This method allows an excessive ring utilization since one slot may carry more than one minipacket within one cycle, depending only on the sender-receiver relations for the communication.

Every minipacket contains the addresses of the receiving and sending ring station as well as an additional reassembly identifier. Therefore, minipackets containing parts of one message can be reordered and the original message assembled. Priority schemes are also possible and - as another advantage - the CS-signalling can be done by minipackets with high priority. Therefore, no separate CS-signalling channel is necessary. Figure 9 elucidates the basic differences of the discussed media access control (MAC) protocols for PS.

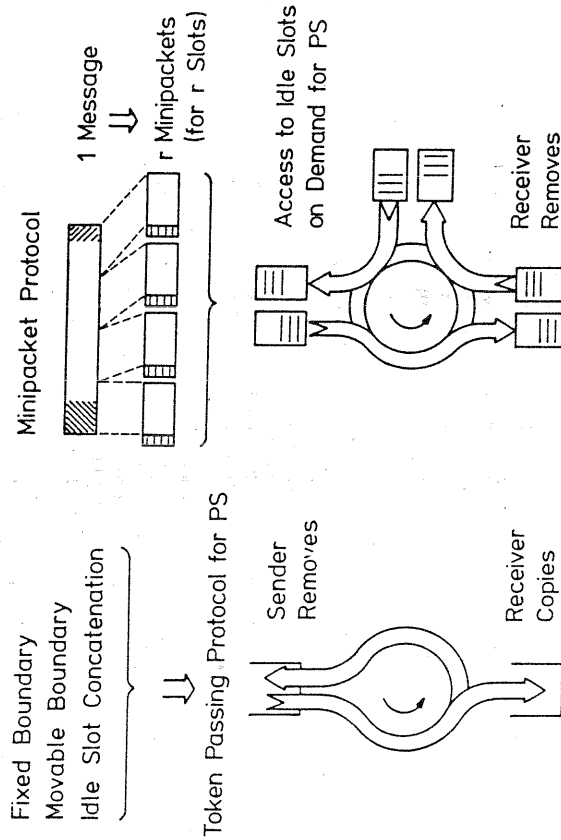


Figure 9: Media Access Control in the Integrated Ring System: Token Passing vs. Minipackets

4.2 Performance comparison

The additional overhead for addressing of minipackets seems to be relatively large. In our laboratory implementation, 16 bits are used for addressing and 48 bits are available for user data. On the other hand, one time slot may be used by more than one minipacket within one frame cycle and the improving of the ring utilization may compensate or even overcompensate for the additional overhead. To answer this question, the performance of such a minipacket system has been investigated [1].

One typical result from simulation runs, comparing all four principles is presented in Figure 10. For a CS/PS-Ring with 10 Mbps and 10 ring stations, the mean waiting times at MAC-layer for a 1 Kbit message are shown by simulation results. The frame consists of 146 time slots, each carrying 64 kbps and 50% of this capacity are assumed to be used by CS-traffic. Each time slot comprises 64 bit user data, in case of a minipacket 16 bit for the minipacket (MAC) overhead and 48 bit user information. This leads to a frame length of 1 ms. The PS-traffic is balanced between all stations in this Figure, but several unsymmetrically loaded systems have been investigated and will be presented in detail in a forthcoming paper. The results are shown versus the total arrival rate of messages at the whole system.

The performance of a system using the partitioned frame principle is relatively bad. Even under a very low PS-load, the mean waiting time for a PS-message is more than 1 msec - the station has to wait for the free token and for the beginning of the PS-part in the frame. The maximum PS-throughput, and also the point of instability in

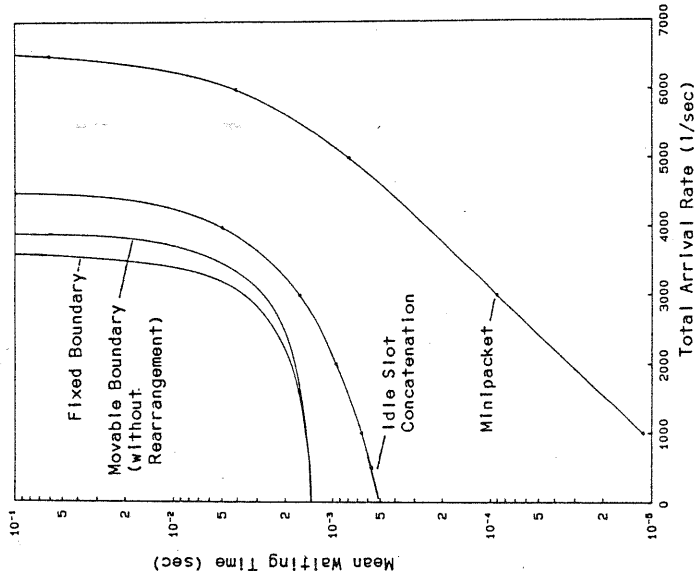


Figure 10: Performance Comparison of the 4 Integration Principles on a Ring System: Mean Waiting Times at MAC-Layer for 1Kbit Messages vs Total Arrival Rate

Figure 10 are reached at a load below 4000 PS-messages per second. In a system with fixed boundary, designed for 73 * 64 kbps full-duplex channels (50% of the total capacity) CS-traffic consumes 88 time slots to achieve 1% probability of loss. In a system with movable boundary but without rearrangement of existing calls, the CS-part comprises only 83 time slots (mean value). These unused slots within the CS-part cannot be used by the PS-traffic and cause this bad performance.

Using ISC as integration principle on the ring system, the performance improves. The total bandwidth here is available for CS or PS, independent from the CS-occupancy pattern. Therefore, the mean waiting time on an empty system results only from the token protocol and a higher maximum load can be carried.

The implementation of the minipacket protocol leads to the fourth curve in Figure 10. Minipackets are sent on demand in every empty time slot, leading to very low waiting times at low PS-load. Removing the minipackets by the receiver doubles the bandwidth available for minipackets. This effect overcompensates the additional addressing overhead within each minipacket, resulting in a very high maximum throughput and also very low waiting times. For more details about the throughput calculation of minipacket-systems see [1].

4.3 Implementation of the CS/PS-Ring

4.3.1 Ring Station

A minipacket ring integrating CS and PS has been implemented at our laboratory. Figure 11a shows the principal structure of a ring station. Several end systems are connected to one ring station, which operates as a cluster controller to reduce costs for the access stations. An end system (e.g. ISDN-terminal equipment, intelligent workstation with telephone and high resolution graphic display or any other computer and communication equipment) is connected to special interface units.

Inside this unit, a line interface realizes the appropriate interface standard, e.g. ISDN-TTs. CS-data are separated from CS-signalling and PS-data, which are directed to the CPU in this unit to perform the signalling protocol as well as the logical link control (LLC) and the segmentation and reassembly (SAR). The SAR partitions a LLC-frame into minipackets with the additional addressing-fields, similar to the new proposed standard IEEE 802.6 [7]. All information from and to the end systems is now adapted to the size of a time slot on the ring. Via the internal data bus all interface units are connected to the media access control (MAC) for PS and the isochronous channel control (ICC) units for CS.

The MAC performs all functions necessary to remove minipackets from the ring, distributes them to the addressed interface unit or inserts minipackets into empty slots. The ICC is controlled by the CS-template, a list of CS-time slots, which are assigned to CS-end systems. The contents of every time slot, marked in that template, is swapped with the contents of the bus interface of the assigned end system.

The ring adapter performs layer 1 functions and delays the transferred signal, allowing to remove minipackets after comparing the addresses. A fast control logic connects this hardware with the internal control bus (VME-bus) and with the station CPU for CS-signalling and additional supervisory functions.

4.3.2 Master Station

Within the ring, one station is needed to generate the synchronous circulating pulse frame and to buffer the frame, compensating different round trip delays. This station, the master station, (Figure 11b) also controls the CS-connections by the CS-

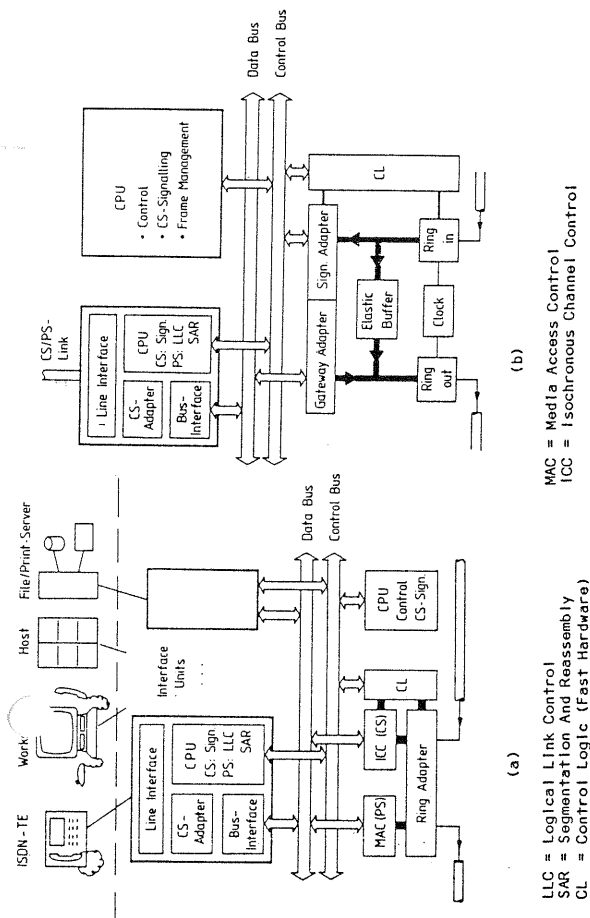


Figure 11: Structure of the Ring Stations:
a) Normal Ring Station Interfacing Various End-Systems
b) Master Station With Gateway to the CS/PS-PBX

signalling and by marking these time slots. It also interfaces the other parts of the inhouse network, using a special interface unit to connect an integrated CS/PS-Link (to an integrated node) and to synchronize the ring with the network. Additional functions are the detection of overload situations and the control of the correct operation of the whole ring system.

5 CONCLUSION

New communication requirements resulting from new services and new multi-functional workstations are claiming for new network solutions, even in the private field. Imbedded into the existing infrastructure a new inhouse communication concept integrating CS with variable and adaptable bandwidth and PS with high throughput rate has been presented.

The concept is based on integrated ring-LANs in smaller spacial areas with higher communication requirements and integrated PBXes substituting today's PBXes. The LANs and the PBXes are interconnected through integrated links.

Several possibilities for the integration within the PBX have been discussed and the implementation of a CS/PS-PBX based on the principle of hybrid switching has been outlined.

The integrated ring-LAN also uses the hybrid switching principle. Different protocols for implementing this switching principle on a synchronously circulating pulse-frame have been presented. Performance evaluation (mean waiting-time for PS-messages) showed that the minipacket-protocol is the most efficient one in spite of the relatively high overhead, necessary for addressing the MPs which will be overcompensated through the multiple utilization of a time slot within one cycle.

6 ACKNOWLEDGEMENT

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7 REFERENCES

- [1] E.-H. Goeldner: An Integrated Circuit/Packet Switching Local Area Network - Performance Analysis and Comparison of Strategies. Computer Networks and ISDN Systems, Vol. 10, Oct. 1985, pp. 211-219
- [2] J.J. Kulzer, W.A. Montgomery: Statistical Switching Architectures for Future Services. ISS'84 Florence, 1984, paper 43A1
- [3] S.R. Amstutz: Burst Switching - An Introduction. IEEE Communications Magazine, Nov. 1983, pp. 36-42
- [4] A.G. Fraser: DATAKIT - A Modular Network for Synchronous and Asynchronous Traffic. ICC'79 Boston, 1979, pp. 20.2.1-20.1.3
- [5] Z.L. Budrikis, A.N. Netravali: A Packet/Circuit Switch. AT&T Bell Laboratories Technical Journal, Vol. 63, Oct. 1984, pp. 1499-1520
- [6] C.J. Jenny, K. Kuenmerle: Distributed Processing Within an Integrated Circuit/Packet Switching Node. IEEE Transactions on Communications, Vol. COM-24, No. 10, Oct. 1976, pp.1089-1100
- [7] Draft of Proposed IEEE Standard 802.6 Metropolitan Area Network (MAN), Media Access Control, Rev. E, Oct. 4, 1985
- [8] R.M. Falconer, J.L. Adams: Carrying Integrated Services on the Orwell Slotted Ring. Int. Seminar on Computer Networking and Performance Evaluation, Tokyo, Sept. 1985, paper 4-1
- [9] I. Mitrani, J.L. Adams, R.M. Falconer: A Modelling Study of the Orwell Ring Protocol. Teletraffic Analysis and Computer Performance Evaluation (North-Holland), 1986, pp. 429-438

HIERARCHISCHE RINGNETZARCHITEKTUREN

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Zusammenfassung

Einen wesentlichen Nachteil großer Ringnetze bildet ihre Unzuverlässigkeit. Dieser kann durch eine hierarchische Struktur behoben werden. Einerseits lassen sich Hierarchien von Ringnetzen aufgrund ihrer Modularität ohne gravierende Störung des laufenden Betriebs erweitern und warten. Andererseits verkürzt ihre Gliederung drastisch die Weglängen und ermöglicht darüber hinaus die Anpassung an vorgegebene räumliche und organisatorische Strukturen. Die Autonomie der einzelnen Subsysteme impliziert die gleichzeitige Übertragung mehrerer Nachrichten im Gesamtsystem. Diese Parallelität bewirkt zusammen mit der Weglängenreduktion eine enormen Leistungssteigerung. In dieser Arbeit werden Kriterien für die Architektur hierarchischer Ringnetze hergeleitet, die ein Optimum an Zuverlässigkeit bieten. Ferner wird gezeigt, daß diese Topologie eine wesentlich größere Leistungsfähigkeit besitzt als einzelne Ringe gleicher Größe und Technologie.

1. Einleitung

Ringe besitzen gegenüber anderen Verbindungsstrukturen erhebliche Vorteile. Alle sendewilligen Stationen erhalten selbst im Hochlastfall in fairer Weise Zugang zum Übertragungsmedium. Fast die gesamte Bandbreite der Ringleitungen kann zur Nachrichtenübertragung genutzt werden, da bei korrektem Arbeiten aller Ringkomponenten keine Übertragungskonflikte auftreten. Darüber hinaus kann garantiert werden, daß jede Station eine Zugriffsmöglichkeit erhält innerhalb eines Zeitintervalls, dessen Länge proportional ist zum Produkt aus der Zahl der Netzstationen und der maximalen Länge einer Nachricht. Diese Zugriffsgarantie ist von entscheidender Bedeutung für Realzeitanwendungen.