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内容简介

本论文选编是西安电子科技大学综合业务网理论与关键技术国家重点实验室，经国家计委、国家教委批准立项以来，所选编的第三本论文集。也是综合业务网理论与关键技术国家重点实验室于1995年10月正式通过国家验收之后所选编的第一本论文集。本论文集选编了实验室研究人员1995年期间在国际刊物，国内核心刊物上公开发表的重要论文或总结报告，共34篇。大部分论文都有一定的创见，从一个侧面反映了实验室在此期间取得的研究成果。内容包括通信网、信源编码及语音图象处理、信道编码与密码等方面。在通信网方面，涉及网络协议及性能分析、ATM技术及无线通信网。在信源编码及语音图象处理方面，主要涉及语音编码、图象编码与数据压缩理论等理论与应用。在信道编码与密码方面有新的调制解调、差错控制及信息的安全保密技术等。

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第 一 部 分

网 络 与 交 换

A New Practical Digital Switching System with Distributed Structure*

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Abstract — This paper presents a new digital switching system with distributed structure. The system is also considered as a communication network composed of some smaller digital PABX stations with ring structure. These PABX stations can be used as the circuit switching terminals of a wide-band integrated services optical fiber ring network or can be constituted as an independent distributed architecture of an exchange.

Key words — Distributed switching, Assignment on demand, Token passing protocol.

I. Introduction

Today, with the rapid development of both the technology and the requirement on communication, how to realize the integrated services digital network (ISDN) becomes more and more important. Some developed countries set up the ISDN experimental prototypes in laboratory in the middle of 1980s and some of them became commodities in late 1980s. As a part of ISDN, the realization of the integrated services local network (ISLN) and to access the voice services to the data communication networks are very important steps toward the ISDN. For this purpose, we devoted to the research project of a large-scale wide-band integrated services fiber optical ring network (i.e., ISFORN) in 1988, and finished the research and development of a new digital switching system with distributed structure in 1991; this switching system is composed of $N(3 \leq N \leq 64)$ small stations, each station has two parts: a digital PABX and a relay interconnecting interface. Because of the function of interfacing, the stations in the switching system could be directly connected to the multi/demultiplexer in node stations of the ISFORN, to provide digital voice services of the circuit switching on the ISFORN.

The system we described in this paper has two advantages. (1) Efficient utilization of channel: The channels to link the stations in the switching system could be assigned on demand with B channel (64kbps) as a basic unit. This is very suitable for digital voice services, but we know, in FDDI-II, the channels could only be divided with 6.312Mbps as a basic unit. (2) The flexibility of the structure: With the distributed structures, the system could even consist of 3 stations up to 64 stations, and all the stations could be concentrated in one room as an exchanger, or could be set as a digital network distributed in an area, and also could become a subsystem of voice services in the ISFORN. In addition, the techniques in the system are valuable for developing the technology of interworking between PABX and LAN or WAN.

II. General Description of ISFORN[1]

The configuration of ISFORN is shown in Fig. 1, where there is an optical fiber ring as the backbone network. To satisfy the requirement for the wide-band digital transmission, it is necessary to use the optical fiber as the transmission medium, and its mono-direction transmission feature is very convenient to make a point to point transmission link

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as a segment of the ring topology. The TDM frame is transmitted on the optical fiber ring with a frame period of 125μs. The bit rate is chosen as 34.368 Mbps, and if needed, this rate could be over than 100 Mbps. To support the integrated services, each frame includes sixteen channels of 2.048 Mbps and a specific packet data group. Besides all kinds of the overhead, the frames are divided by two parts: packet switching field and circuit switching field, that is, the hybrid switching mode with the circuit switching combining the packet switching is adopted. The frame format is shown in Fig. 2. According to the overall scheme of the ISFORN, the switching system shares 4 pre-assigned isochronous channels of 2.048 Mbps in the circuit switching part. So, the transmission bit rate on the interface between the nodes of the optical fiber ring network and the PABX station is 8.192 Mbps, and this interface is implemented by the multiplexer and demultiplexer in the node of the ISFORN.

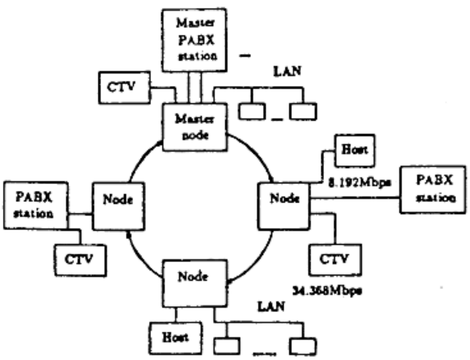

Fig. 1. The architecture of the ISFORN

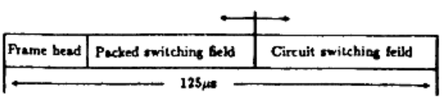

Fig. 2. The TDM frame structure of the ISFORN

III Realization of the Switching System[1-3]

1. The overall scheme

In the overall structure of ISFORN shown in Fig. 1, if all the PABX stations are connected with a 8.192Mbps channel corresponding to the interconnecting trunks, an independent switching system will be formed as shown in Fig. 3, where the system is a digital communication ring network. The maximum distance between neighbouring two stations is 1.5 km. The bit rate on the trunks is 8.192Mbps that include 128B (1B=64kbps) channels. Among them, 120B channels can be used for supporting the circuit switching service and be shared with assignment on demand by the users of all the stations in this switching system; 4B channels are used as the common signalling channel, and other 4B channels are used for frame synchronization. The frame structure is shown in Fig. 4, in which, the four time slots (ts0) will be used to transmit 32 bit Generalized Barker Code as the unique code for the frame synchronization. The four time slots (ts16) become a $4 \times 64 = 256$ kbps channel, which can be used as a common signalling channel, through which all stations will compose logically a sub-ring network to perform the signalling transfer of the switching system.

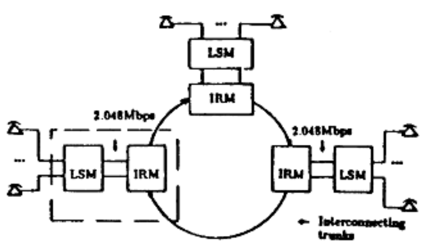

Fig. 3. The architecture of the switching system

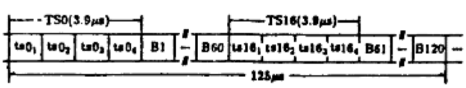

Fig. 4. The relay frame structure of the switching system

2. The signalling protocol

The signalling frame structure on the sub-ring is coincident with that of HDLC and the token passing protocol is employed. The signalling frame and the token frame circulated on the sub-ring are distinguished according to the control octet in the HDLC frame. The token is a license for each station that can send signalling. After initiating of the subring, the token will be generated from the master station, and when the next station has received the token, it will first send out the signalling frame and then send out the token frame to the following next station if it has some signallings to send; or it will send out only token frame immediately to the following next station if it has no signalling needed to send. In this way, the token will be passing from one station to the next and so on along the ring. During the process of the token passing, the master station is responsible for the maintenance and management of the token. There is a timer in the master station to measure the real time (TRT) for the token circulating one cycle around the ring. If the TRT is beyond the upper limit, it can be considered as the token has been lost during the transmission process. In this case the master station will generate a new token. If TRT is less than a lower-limited value, the received token can be considered as a false one and will be deleted.

The assignment on demand and management of the 120B channels will be executed by the master station also. There is a busy/idle sign table of channels to be set up in the master station. When there arrives a request of setting up a connection for communication between the users of two stations, the calling party must make an appeal to the master station for one B channel and inform the master station the calling and the called address or telephone number by a signalling. Then the master station will select an idle B channel to them according to the mark of the busy/idle table (in the ring network, only one B channel is needed for the duplex communication of two users), and at the same time, set the idle bit to busy in the table. At the end of the communication, the calling user or the called user will send out a signalling to request the master station to release the used channel. The master station will disconnect the link between calling and called party and then change the busy/idle table again to reset that channel to be idle. At last, the master station will transmit a signalling to announce that the channel is released completely.

3. The hardware of the switching system[4,5]

The hardware block diagram of each station of the switching system is shown in Fig. 5. Each station can be divided into two parts: the local switching module (LSM) and the interconnecting relay module (IRM). The LSM is a common time division switch, which mainly consists of line interface unit and time slot exchange to realize the local switching function. Also, it must have an interface with the IRM to transfer the control information mutually. The IRM is responsible for the user information and signalling message transferred among the interconnecting PABX stations on the ring network. That is, it must perform the functions such as synchronization, transmitting/receiving and processing the signallings, receiving/transmitting the user's digital information from or to the ring network and transferring the digital information that is not destined for this station. The 2.048 Mbps PCM time division multiplexing highway is adopted for the physical interface between the LSM and the IRM.

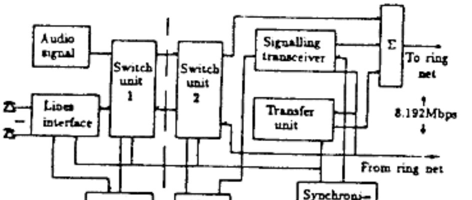

Fig. 5. The hardware block diagram of a station of the switching system

4. The software[5]

Each station of the switching system contains two microprocessors. They carry out the control functions of the LSM and the IRM individually. The control procedure of the LSM can be divided into two levels: the interrupting service level and the basic level. The basic level of the routine will be executed repeatdly after the PABX is powered on or is reset. For the interrupting service level, the interrupt signal is generated by the hardware with the period of 8 ms and the executed periods of each interrupting service routine are all the integral times of 8 ms. The software of the IRM is composed of a main routine, many subroutines and interrupting service programs. Each interrupting service program is started by one interrupting source. The main routine is in charge of coordinating the interrupting service programs and all subroutines.

IV. Conclusion

We have realized the experimental prototype of the switching system with three stations as shown in Fig. 3 and have finished the communication testing for all stations being connected into the optical fiber ring network as shown in Fig. 1. which works well and reliably. The features of this digital switching system are as follows:

• To realize the combination of the technology of the high speed ring network and PABX. it is the first in China;

• To implement the assignment on demand of the channel capacity with the B channel as a basic unit and only need one B channel for the duplex communication between two users, which raises utilization of the channels;

• To offer a distributed architecture of SPC exchange, which connects many smaller capacity exchanges or switching modules using a high speed local area network.

Of course, many supplement functions of this exchanger would be developed and many works should be done to improve the reliability and make it practicable.

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流体神经网络模型用于通信网络的路径选择*

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摘要　流体神经网络是一种能直观描述流体流动物理性质的神经网络模型. 文中将通信网络对应于一个流体神经网络，从而给出了一种通信网络路径选择的并行算法. 模拟结果表明，这一算法能快速找到最佳网络路径.

关键词：路径选择；流体神经网络；最大流量

中图分类号：TN913.2；TP393.2

网络路径选择是将源端发出的信息经最小代价路径传送到目的端. 这一代价可以是路径最短、延迟最小、经过的中间结点数或链路数最少等具体的物理量. 路径选择问题是网络通信领域中的一个重要问题.

以往的路径选择算法主要依据图或树上的串行搜索[1]. 神经网络和神经计算机的出现使大规模并行计算和处理成为可能. 由于并行计算能极大地提高运算速度，因此许多研究人员尝试用神经网络解决路径选择问题[2]. 通常运用神经网络方法的主要思想是，先构造出对应于路径代价的能量函数，然后神经网络按能量下降方式迭代，收敛至最小值（稳定值），神经网络的状态就给出了最佳路径的解. 这种神经网络的神经元数目常常要多于通信网节点数的好几倍，且能量函数有多个约束项，从而使神经网络结构很复杂.

文中基于流体神经网络模型给出了一种网络路径选择算法，这种方法不涉及能量函数，而是直接从流体神经网络的物理性质出发，来完成网络路径选择.

1　流体神经网络

Guo 基于 Hopfield 连续神经网络模型[3]给出了一种具有直观流体性质的流体神经网络模型[4]，在这种流体神经网络中，每个神经元可看成是一个盛 u 流体的容器，容器内流体的高度表示该神经元的状态，连接权表示容器间的连通通道，权值的大小表示两个容器间的流体流通能力. 每个神经元的数学特性描述为微分方程

$$\frac{\mathrm{d}u_i}{\mathrm{d}t} = \sum_j T_{ij}(S_j - S_i) - 2|T_{ii}|u_i + I_i \tag{1}$$

其中　$S_i = g(u_i)$、g 为非线性函数，$g' > 0$。$T_{ij} = T_{ji} \geq 0, i \neq j$；$T_{ii} = -\sum_{j \neq i} T_{ij}$

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在此神经系统中，u_i 为第 i 个容器内 u 流体的体积，S_i 为容器内流体高度，函数 g 刻划了容器的形状，$-2|T_{ii}|u_i$ 表示容器的漏量，I_i 是输入流速. T_{ij}表示容器 i 与容器 j 的连通性，$T_{ij}(S_j-S_i)$是单位时间内从容器 j 流入容器 i 的流量.

系统总体积定义为

$$U=\sum_{i=1}^{N}u_i$$

不考虑漏项，方程(1)可写为

$$\frac{\mathrm{d}u_i}{\mathrm{d}t}=\sum_{j=1}^{N}T_{ij}S_j+I_i \tag{2a}$$

$$S_i=g(u_i),\quad T_{ij}=T_{ji}\geqslant 0\ (i\neq j)$$

$$T_{ii}=-\sum_{j\neq i}T_{ij} \tag{2b}$$

当系统输入输出相等时，即 $\sum I_i=0$，系统总容量不变，对此Guo证明了流体神经网络必能达到稳定，且稳定点唯一[4]. 从直观的物理意义上来看，流体神经网络的稳定实际上是维持流体的动态平衡，也就是使每个容器的流入量与流出量相等. 稳定时容器之间的液面高度差指示了流体流动的方向.

2 流体神经网络路径选择算法

一个通信网络可以用节点和链路构成的图来描述，一个节点代表一个通信单元. 节点间的链路代表节点间通信的代价，如距离、延迟、发送每比特(bit)信息所用的时间等. 表示这样一个图可以引入连接矩阵 C，其中元素 c_{ij}表示节点 i 到节点 j 的代价. 由于计算机通信网中两节点相互通信使用同一信道，因此有 $c_{ij}=c_{ji}$，即 C 为对称的. 矩阵中用∞表示两节点间无直接链路. 由于不考虑节点到其自身的连接，所以令对角线元素 $c_{ii}=\infty$.

对比流体神经网络和通信网络，可以发现在直观的物理意义上，它们之间有很大的相似性. 信息包从源节点经通信网发送到目的节点，可以看作是一定量的流体流入一个容器，神经网又从另一个容器流出. 流体流动自然遵循最速下降的原则，这一原则同样可以作为路径选择所遵循的原则.

将流体神经网络对应于通信网络，每个神经元对应一个通信单元，神经元的连接对应通信网单元之间的链路. 由于流体流动时尽可能通过连通性好的通道，而信息传递尽可能经过代价小的路径. 因此代价越小，连通性越好. 反之，代价越大，连通性越差，没有直接链路对应连通性为 0. 令第 i 个和第 j 个神经元的连接权值为 T_{ij}，则有

$$T_{ij}=\begin{cases}0 & c_{ij}=\infty\\ c_{ij}^{-\alpha} & c_{ij}\neq\infty,\quad 其中\ \alpha>0\end{cases} \tag{3}$$

最简单的情况可取 $\alpha=1$.

流体神经网络稳定时，两个神经元之间的流量为一定值. 在一个神经元与其连接的各条通道中，流出量最大的通道从概率意义上讲是流体自然选择的最可能流出的通道. 因此直观上从源节点出发，经流量最大的通道到达目的节点的路径是一条代价最小或接近最小的路径.

由上面的讨论，可以得到以下路径选择算法 FNNR.

Step 1 构造一一对应于 n 节点通信网络的有 n 个神经元的流体神经网络，每个神经元对应一个通信节点，神经元连接权值与通信节点间代价的关系满足式(3). 令 $S_1, \cdots, S_n$ 为容器液面高度，$u_1, \cdots, u_n$ 为容器内流体体积，它们满足关系 $S_i = g(u_i)$，g 为非线性函数，且 $g' > 0$. 设初始时刻，$S_1(0) = \cdots = S_n(0) = S_0$，$u_1(0) = \cdots = u_n(0) = u_0$.

Step 2 对应源节点 S_1 和目的节点 S_n 分别加输入和输出，且出入量相等，设为 I $(I > 0)$. 神经网络开始并行迭代，每一种神经元满足微分方程

$$\frac{\mathrm{d}u_i}{\mathrm{d}t} = \sum_{j=1}^{N} T_{ij}(S_j - S_i) + I_i \tag{4}$$

其中

$$I_i = \begin{cases} I & i \text{ 为源节点} \\ -I & i \text{ 为目的节点} \\ 0 & \text{其它节点} \end{cases}$$

Step 3 网络迭代至稳定. 从 S_1 出发搜索最大流量通道至神经元 S_{i1}，再从 S_{i1} 出发经最大流量通道至 S_{i2}，如此反复直至 S_n.

Step 4 输出点列 $S_1 \to S_{i1} \to S_{i2} \to \cdots \to S_n$ 作为所选的路径.

当任一条链路上的代价都相等时，即代价矩阵 C 有

$$C_{ij} = \begin{cases} 1 & i,\ j \text{ 之间有链路} \\ \infty & i,\ j \text{ 之间无直接链路} \end{cases}$$

此时最小代价路径就是经过最少节点数或链路数的路径. 因为相互链接的两节点 i 和 j，j 容器流入 i 容器的流量是

$$T_{ij}(S_j - S_i) = \frac{1}{C_{ij}}(S_j - S_i) = S_j - S_i$$

它等于 j 容器与 i 容器的液面差，所以 Step 3 中搜索最大流量通道可以简化为寻找最速下降的通道.

3 模型实验结果

图 1 所示是一个 16 节点的通信网(录自文献[2]). 其代价矩阵如表 1 所示.

表 1 代价矩阵 C

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
1	∞	1	1	1/8	∞	∞	∞	∞	∞	∞	∞	∞	∞	∞	∞	∞
2	1	∞	1	∞	1	1	∞	∞	∞	∞	∞	∞	∞	∞	∞	∞
3	1	1	∞	1	1	∞	∞	∞	∞	∞	∞	∞	∞	∞	∞	∞
4	1/8	∞	1	∞	1	∞	∞	∞	1	∞	∞	∞	∞	∞	∞	∞
5	∞	1	1	1	∞	∞	∞	1	∞	∞	∞	∞	∞	∞	∞	∞
6	∞	1	∞	∞	∞	∞	1	1	∞	∞	∞	∞	∞	∞	∞	∞
7	∞	∞	∞	∞	∞	1	∞	1	∞	∞	1	1	∞	∞	∞	∞
8	∞	∞	∞	∞	1	1	1	∞	1	1	1	∞	∞	∞	∞	∞
9	∞	∞	∞	1	∞	∞	∞	1	∞	1	∞	∞	∞	∞	1/4	1/4
10	∞	∞	∞	∞	∞	∞	∞	1	1	∞	1	∞	∞	1	1	∞
11	∞	∞	∞	∞	∞	∞	1	1	∞	1	∞	1	1	∞	∞	∞
12	∞	∞	∞	∞	∞	∞	1	∞	∞	∞	1	∞	1	∞	∞	∞
13	∞	∞	∞	∞	∞	∞	∞	∞	∞	∞	1	1	∞	1	∞	1
14	∞	∞	∞	∞	∞	∞	∞	∞	∞	1	∞	∞	1	∞	1	1
15	∞	∞	∞	∞	∞	∞	∞	∞	1/4	1	∞	∞	∞	1	∞	1/2
16	∞	∞	∞	∞	∞	∞	∞	∞	1/4	∞	∞	∞	1	1	1/2	∞

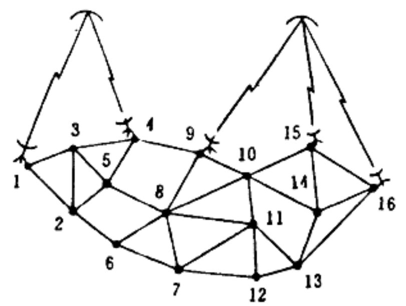

图 1 16 节点通信网

从观察可以发现节点 1 和节点 13 之间通信所经最小代价路径是

$$1 \rightarrow 4 \rightarrow 9 \rightarrow 16 \rightarrow 13$$

在 IBM PC 微机上对图 1 所示网络按表 1 及式(4)应用流体神经网络算法得到的模拟结果表明，仅用 4 次并行迭代和 64 次串行比较就找到了最小代价路径，计算量远远低于文献[2]中算法所用的 250 次迭代.

实验中发现，连通性和系统的输入输出流速对收敛速度有很大影响. 从表 2 和表 3 中可以看出，连通矩阵 T 成倍增大，表明连通性越强，收敛速度则越快，但神经元间的状态变化越不明显. 输入输出流速 I 越大，系统达到稳定所需时间越长. 状态变化反映了流向，选择路径需要神经元状态间有明显的变化. 因此固定连通矩阵在保证变化的情况下，调整输入和输出流速可以加快收敛.

表 2 联接权对稳定状态和迭代次数的影响

联接权	稳定状态				迭代次数
T	62.2	60.2	60.2	60.9	10
	60.1	60.0	60.0	60.0	
	60.0	60.0	59.8	59.8	
	57.2	59.8	60.0	59.8	
$2T$	61.2	60.1	60.1	60.6	6
	60.0	60.0	60.0	60.0	
	60.0	60.0	59.9	59.9	
	58.4	59.9	60.0	59.9	
$4T$	60.0	60.1	60.1	60.4	4
	60.0	60.0	60.0	60.0	
	60.0	60.0	59.9	59.9	
	59.0	59.9	60.0	59.9	

表 3 输入/输出流速对稳定状态和迭代次数的影响

输入/输出流速	稳定状态				迭代次数
4.5	60.0	60.1	60.1	60.4	4
	60.0	60.0	60.0	60.0	
	60.0	60.0	59.9	59.9	
	59.0	59.9	60.0	59.9	
9.8	63.7	61.6	62.1	62.9	23
	61.4	60.2	59.3	59.8	
	59.6	59.3	58.7	58.3	
	56.0	58.6	59.4	59.0	
10.0	69.1	64.8	66.1	67.4	40
	64.2	60.6	58.0	59.3	
	58.7	57.8	56.4	55.3	
	50.8	56.1	58.0	57.3	

4 结束语

文中应用流体神经网络模型给出了一种通信网络路径选择算法. 这种方法并不是依据设定能量函数下降至最小来求解最优路径的，这一点不同于 Hopfield 求解 TSP 问题的神经网络算法[5]以及 Rouch 和 Winarske 的神经网络路径选择算法[2]. 这一算法是将流体神经网络直接对应于通信网络，把通信网内信息的传送类比于神经网络内的流体流动，利用流体流动的物理性质实现网络的路径选择. 这种方法简便直观，使用神经元数目及迭代次数远低于 Hopfield 网络[2]. 模拟实验结果表明这一方法能有效而迅速地完成路径选择.

这一方法仍涉及一些进一步的问题有待解决：

(1) 虽然模拟结果显示 FNNR 算法能够给出最优路径，但在理论上还有待证明是否所得结果即为最优解，或在多大程度上接近最优解.

(2) 神经元非线性函数的选取影响到流体神经网络的收敛性，选择什么样的非线性函数值得进一步研究.

(3) 具有不可靠单元的通信网路径选择是一个重要问题[6]，如何运用流体神经网络解决这一问题将是一个很有价值的问题.

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A fluid neural network for routing communication traffic

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Abstract

A fluid neural network is a kind of neural computational model that has visual properties of describing the way of fluid flow. This paper presents a parallel algorithm for routing a communication network by comparing a communication network to a fluid neural network. Simulation results indicate that the routing algorithm can find the optimal path very fast.

Key Words: path routing; fluid neural network; maximum flow

无冲突分组预约多址接入(CF-PRMA)协议及其性能*

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摘要 文中指出了分组预约多址接入(PRMA)协议的一些缺点，并提出在分组预约多址协议帧结构的基础上，增加少量的信令、控制时隙，实现一种无冲突的分组预约多址接入(CF-PRMA)协议，理论分析与计算机仿真均表明，CF-PRMA 比 PRMA 的性能有明显的提高.

关键词：个人通信网(PCN)；多址接入协议；分组话音
中图分类号：TN913.24

近年来，作为第三代移动通信系统的个人通信网(PCN)正受到人们越来越多的重视；而以微蜂房通信网为基础、传输数字化话音的蜂房分组交换(CPS)系统是实现个人通信网的一种引人注目的方法. 在微蜂房分组交换系统中，基站与其所服务的通信终端形成一个星型的网络结构；为了使分散的各通信终端能利用共享信道与同一基站进行通信，Goodman 等人提出了一种分组预约多址接入(PRMA)协议. 简单地说，PRMA 是预约 ALOHA 与 TDMA 的结合，它首先利用 ALOHA 竞争空闲时隙，竞争成功后，将唯一拥有一个预约时隙传送后续的分组，已预约的通信终端以 TDMA 方式共享信道. 但标准 PRMA 存在着一定的局限性.

首先，PRMA 需要在话音有声期的开始，通过竞争来预约一个时隙. 由于预约具有随机性，所以各通信终端之间会产生冲突，从而降低整个系统的容量. 其次，PRMA 不能有效地控制各通信终端话音的时延特性，因而会造成有的话音分组因时延过长而被丢弃，与此同时有的话音分组却没有时延或时延较小. 再者，对于一段话音来说，各话音分组对重建话音的质量所起的作用并不相同. 对于汉语话音来说，话音段开始部分的话音分组(清音和过渡音)比中间及末尾部分的话音分组(浊音)更重要些. 对于 PRMA 协议，话音分组的丢失总是发生在话音段的开始部分，也就是说，总是重要的话音分组被丢弃. 另外，在 PRMA 协议中，话音分组可能是连续被丢弃的，这不利于在接收端对丢弃的话音分组进行有效的恢复.

为了克服 PRMA 的这些缺点，文中提出了一种无冲突分组预约多址接入(CF-PRMA)协议. 该协议通过增加少量的信令、控制时隙，从而消除 PRMA 中的冲突现象，同时可以有选择地丢弃话音分组(如只丢弃不太重要的话音分组)从而使话音分组的丢弃对接收端重建话音的质量影响最小.

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1 无冲突分组预约多址接入(CF-PRMA)协议

与PRMA协议相同，CF-PRMA协议控制上行的信息以便使分散的各通信终端共享一个到达基站的信道. 传输以帧为单位，每帧由许多时隙组成，并且假定帧速率等于话音分组的到达速率.

在CF-PRMA协议中，每个上行的帧内包括3种不同的时隙：信令时隙、控制时隙和话音时隙. 具体的帧结构见图1.

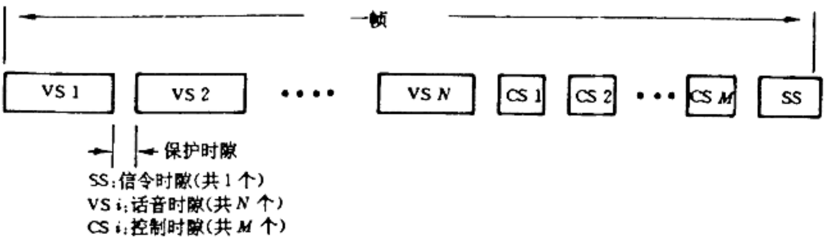

图1 CF-PRMA协议的帧结构

CF-PRMA协议的工作过程如下：

在某一通信终端需要进行通话时，首先在信令时隙(SS)发出建立通话链路的请求，发送的信息包括本终端的号码、对方终端的号码等. 当基站成功地收到这一请求后，根据系统内通话终端的数目等信息，决定是否接受通话请求，如果接受则给这一终端唯一分配一个控制时隙(CS_i)，并通过下行信道通知此通信终端. 这样，处于通话状态的各终端都拥有各自不同的控制时隙.

当通话终端处于有声期时，对其相应的控制时隙置位；处于无声期时，对其相应的控制时隙复位. 最简单的情形是：控制时隙只有1比特，有声期时控制时隙发送“1”，无声期时发送“0”或不发任何信息. 当基站得知通话终端处于有声期时，将给此终端分配一个或几个话音时隙；如果话音时隙数少于处于有声期的终端数，则首先给话音分组时延长的终端分配话音时隙.

如果没有空闲的话音时隙、且所有终端分组话音的时延都达到了极限值(D_{max})，则丢弃话音分组是不可避免的. 在作者所设计的CF-PRMA协议中，可以有选择地丢弃那些不重要的话音分组. 对于汉语音来说，可以首先丢弃处于浊音段的话音分组，这样对接收端的重建话音质量影响也最小. 同时，如果浊音段的话音分组被丢弃，在接收端还可以根据话音信号的相关性，进行有效的恢复. 研究表明，增加有效的话音恢复技术，可接受的话音分组丢弃率可以达到8%～10%；而清音段因其相关性较弱，话音恢复技术不太有效.

通话结束后，通话终端在信令时隙发出拆除通话链路的信息，之后释放相应的控制时隙，至此完成了整个通话过程.