PROPERTIES OF A TDMA PICO CELLULAR OFFICE COMMUNICATION SYSTEM

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ABSTRACT

CEPT has decided to base the new standard for European Digital Cordless Telecommunications, DECT, on TDMA/TDD, intented for versatile residential, office and public telepoint applications.

Properties of a TDMA/TDD micro or rather pico cellular cordless telephone system using decentralized dynamic channel allocation and hand over are presented. The system has 16 or 8 time division time duplex channels (slots) per carrier, each occupying 2 MHz or 1 MHz. The TDMA frame is 16 ms long and each slot has bits for synchronization, signalling and data. The data speed is suited for 32 kbps speech codecs. Procedures have been developed for efficient and quick dynamic channel allocation and hand over, using the fact that one single radio can simultaneously monitor or communicate on all 16 or 8 time slot channels. Simulations, for a specific six storeyed building indicate that 16 channels can provide wireless communication for up to 30 % of the telephones, and 32 channels for up to 100 % of the telephones. Slot synchronization between adjacent systems is desirable, but not necessary. A modest echo control is needed on the trunk level interface to the PABX or PSTN, if 2 wire transmission is used. Tests show that no time dispersion equalizers are needed, and that antenna diversity is effective against fading dips and time dispersion.

INTRODUCTION

In January 1988 CEPT (the standardization body for the European PTT's) decided to base the standard for medium and long term applications of advanced Digital European Cordless Telecommunications, DECT, on Time Division Multiple Access/Time Division Duplex, TDMA/TDD, (1).

The possible applications range from PABX with wireless extensions, wireless PABX, wireless key systems and residential telephone systems to telepoints. See Fig. 1.

The radio interface specification by CEPT is suggested to consist of two levels (6).

Level 1 is a mandatory minimum specification that secures co-existence on the same band of TDMA systems from different manufacturers. E.g. the frame cycle time has to be common for all systems.

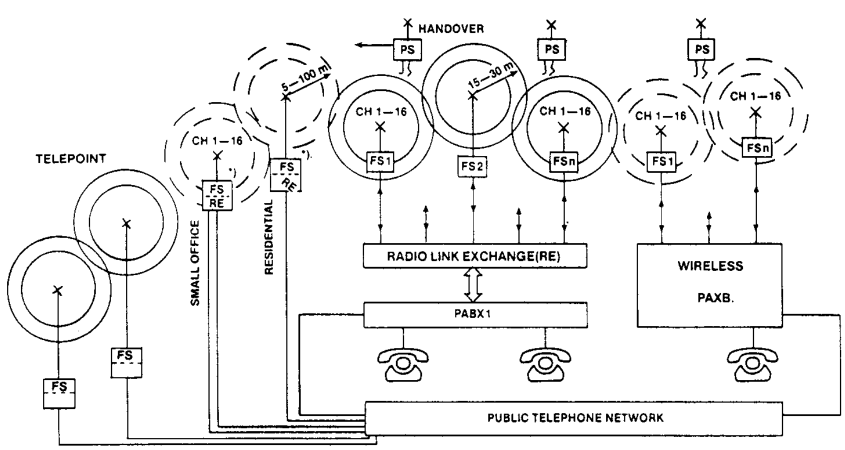
This will allow freedom for private (business) systems to modify the air interface etc. according to changes in needs, cost and technology as long as the co-existence is not adversly effected. E.g. introduce lower bit codecs, data services etc.

Level 2 is an optional detailed air interface specification (speech service) needed for public (Pan-European) telepoint applications.

The TDMA frame will operate on a (low) number of radio carriers on frequencies common for Europe around 1.6-17 GHz. First products are expected on the market in early 1990'ies.

This paper presents properties of a TDMA/TDD pico cellular cordless telephone system concept. The concept has been proposed to CEPT by the Swedish PTT (2) and ESPA (European Manufacturers of Pocket Communication Systems) (3) as a base for DECT.

The studies have been made on the 900 MHz band, but can with some care also be used to estimate the behaviour on 1.6 GHz, where the main allocation of DECT will be.



*) No need for synchronization.

FIG. 1. PROPOSAL DECT-SYSTEM.

It is very essential to avoid traditional frequency planning and channel allocation for individual (private) systems.

The solution is Dynamic Channel Allocation, which avoids frequency planning for individual systems and permits system sales on a general (not individual) license.

The market for business office systems is regarded as the most important. TDMA is very well suited for pico cellular office systems using decentralized dynamic channel allocation and hand over.

The Swedish PTT will soon introduce an interim TDMA/TDD cordless telecommunication service (2) on the 880 MHz band, and later extend the service to 1.6 GHz when the DECT specification is finalized.

SYSTEM DESCRIPTION

The system has 16 time duplex channels (time slots) on 2 MHz wide frequency carriers. For simplicity we describe the single carrier case with 16 time slots. Fig. 2 shows an example of the TDMA frame and of the traffic slot. This frame allows the use of 32 kbps speech codecs.

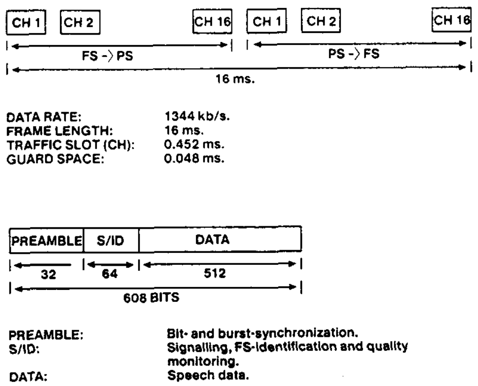


FIG. 2. TDMA FRAME AND TRAFFIC SLOT.

The fixed stations (FS) at each pico will be inexpensive. The reason is that TDMA with one transceiver can communicate simultaneously on all 16 channels.

Dynamic channel allocation

For the office case each FS is all times active on at least one slot. In idle state, a portable (PS) is locked to the closest (strongest field strength) FS, and checks for a paging signal in the S/ID part available in all active slots. Special signalling slots are not introduced.

The channel (time slot) allocation is dynamic and decentralized to each pico cell and each portable (PS). The PS identifies the strongest FS by reading the FS identification code in the S/ID part of each slot. When a speech channel is wanted the portable communicates with the least interference at the position of both the PS and of the FS. Thus there is no need for central control.

Since all the stations (also the PS at their positions) have continous information of the status on all channels, the dynamic channel allocation can be made very efficient and fast. Also handover from one FS to another in the same office, can be made very efficient, fast and with no interruption of the speech. The portable can communicate with the old FS on one time slot, while building up the communication to the new FS on another slot.

The same speech channel (time slot) can be reused within the same office. The larger the office, the more cells, the more reuse and the higher the capacity.

The range is C/I limited. Thus the capacity can be increased by installing the FS closer and closer.

It is not only possible to reuse the channels (slots) within a system. Different level 1 systems can use the same radio frequency channel, without any frequency planning. The system does not care if the adjacent cells belong to the own system or not.

Substantial signalling capacity in each slot

The preamble and the S/ID part of each slot make it possible for the FS and the PS to resynchronize and identify the transmitting station or receive a message from one single slot. This seems to be a key feature for good protection against time dispersion, for quick effective dynamic channel allocation with hand over, for good power saving procedures for the portables and for effective control of the switched antenna diversity employed at the fixed stations.

Other features

The system performs very well with antenna diversity and needs no time dispersion equalisation. See Appendix.

The concept allows adjacent systems to co-exist without need for common frame synchronization, as long as the frame clocks are reasonably stable. See Appendix.

A modest echo control is needed on each trunk to the PABX or PSTN if 2-wire connection is used. Full 4-wire connection needs no echo control. See Appendix.

TRAFFIC CAPACITY

Simulations have been made for a pico cellular office system in a six storeyed modern building with a floor plan according to Fig. 3. Nine or fourteen fixed stations (FS) per floor are placed as the x and o indicate.

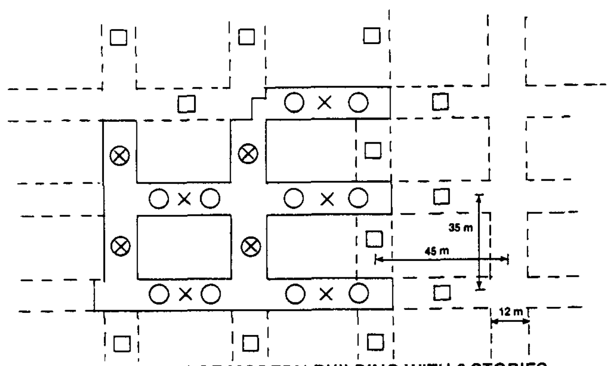


FIG. 3. FLOOR PLAN OF MODERN BUILDING WITH 6 STORIES.

The propagation model

Fig. 4 shows the propagation model used. The two curves indicate the space for a 20 dB shadow variation with even distribution. Rayleigh distribution is added. The model is derived from propagation measurements in the building. The measurements were made with the FS transmitter location in the middle of corridors and the receiver in rooms along the corridor or along other corridors on the same storey and on other storeys. The model covers statistics from a line along a corridor as well as 45 degrees across the house on the same storey. The added isolation between the floors was in average at least 15 dB except for rooms that were located parallell across the open space between the houses. To compensate for that the added isolation between the floors was set to 10 dB. In the vertical direction all channels can be reused (at least in average) every three storeys. Thus the simulation is made for 3 storeys only.

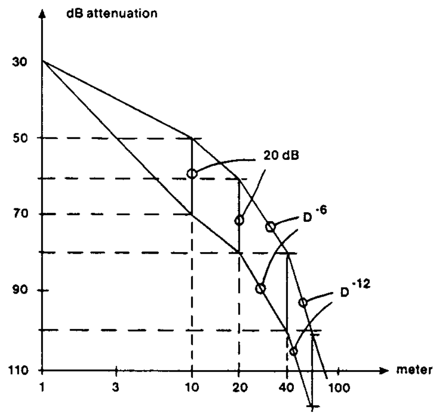


FIG. 4. THE PROPAGATION MODEL.

Telephone density and traffic presumptions

Total traffic for 3 storeys: 100 E
Traffic per storey: 33.3 E
Telephones per storey (0.1 E per telephone): 333
Total area per storey: 4000 m^2
Telephone density: 1/12 m^2
Traffic density per 3 storeys: 12 500 E/km^2
(50 % built)

The simulation is made for 10 E, 30 E, 70 E and 100 E, corresponding to 10 %, 30 %, 70 % or 100 % of the telephones being wireless.

Call length: Average 100 sec. Exponential distribution.
Call interval: Average 4 calls per tel and hour. Poisson distribution.

Radio characteristics

21 dB C/N corresponds to 89 dB attenuation in the propagation model, Fig. 4.

Antenna diversity is used at the FS.

Test system

A small scale test system with 2 mobiles and up to 5 FS has been built September 1987. It is connected to a PABX and uses decentralized dynamic channel allocation with hand over, FS switched antenna diversity and no time dispersion equalizers. The system offers duplex speech service and functions very well.

Procedures

The procedures used in the simulations are very similar to those we are testing in the test system.

The portables are evenly distributed over the three floors. The portables are randomly selected for calls.

For a call set up the portable chooses the strongest FS and then the channel is chosen that has the lowest field strength both at the FS and at the portable unit.

The interferring field strength to the portable is added up from all the FS that transmit on the chosen channel. And the interferring field strength to the FS is added up from all portables (in their actual positions) that transmit on the chosen channel.

The value of the field strength is calculated by deciding the distance between the receiver and the transmitter and for this distance taking a random attenuation sample within the 20 dB (shadowing) window of the propagation model. From this attenuation the average received field strength is derived, and for that two random samples of the Rayleigh distribution are taken, and the best sample is chosen.

For a call set up C/I (or C/N) has to be better than 21 dB. This means that there is only 1 % risk that a slowly moving portable recives less than 11 dB C/I. A call is terminated if C/I (C/N) is below 11 dB for more than 10 sec.

For every new call set up and call termination, the C/I is recalculated for all active receivers, and if C/I is below 21 dB, a channel change and perhaps base change is initiated and if needed repeated.

In the table below blocked means that a call set up does not succeed, and interrupted means that an ongoing telephone call, by a new call set up is forced to change channel, but does not succeed to find a new channel.

Each of the 11 simulation runs contains 10 000 calls. Thus 0.01 % corresponds to 1 call.

The total number of channels are 16 or 32. For 32 channels 4 MHz are needed. The portables can scan all 32 channels.

Run no	No. of chan.	Traffic % or E	C/I mean	No of FS	Blocked calls	Interrupted calls	Average no of used ch/FS	No of wireless telephones	No of channels per FS to give 0.5 % G.O.S.
1	16	10	53dB	27	-	-	0.4	100	3
2	16	30	42dB	27	1.4 %	0.25 %	1.1	300	5
2b	2x8	30		27	1.5 %	0.3 %	1.1	300	5
3	16	30	45dB	42	0.2 %	0.01 %	0.7	300	4
3b	2x8	30		42	0.2 %	0.02 %	0.7	300	4
4	32	70	45dB	27	0.7 %	0.04 %	2.6	700	8
4b	2x16	70		27	0.8 %	0.1 %	2.6	700	8
4c	4x8	70		27	0.9 %	0.2 %	2.6	700	8
5	32	70	48dB	42	0.01 %	-	1.7	700	6
5b	2x16	70		42	-	-	1.7	700	6
6	32	100	46dB	42	0.1 %	0.01 %	2.4	1000	8
6b	2x16	100		42	0.1 %	-	2.4	1000	8
6c	4x8	100		42	0.2 %	0.04 %	2.4	1000	8

Table. Grade of service as function of traffic and other parameters.

The simulations are made for 32, 16 and 8 duplex time slot channels per carrier. The corresponding bandwidth per carrier is 4, 2 and 1 MHz. 2 carriers are introduced in the b runs and 4 in the c runs, with the limitation that each FS can operate simultaneously only with one channel per time slot position.

The traffic and the number of FS is for three storeys. The number of wireless telephones is ten times the traffic E.

Comments to the results in the Table

The simulation indicates that for one 16 channel system up to 30 % of all telephones can be wireless, and that all can be wireless if a 32 channel system is used, when a 0.1 to 0.2 % blocking limit is used.

The average number of used channels per FS station is derived by dividing the total traffic in E by the total number of FS stations.

It is interesting to note that the average number of used channels is less than 1 for the 16 channel system and about 2.5 for the 32 channel system.

The FS that is simulated in runs b and c can be implemented with one radio only, that can shift carrier between time slots, by a quick switch between local oscillators or between two simple (slow) synthesizers. The result shows hardly any penalty compared to the runs with full access to all channels at any time. The reason is of course that in average only about one slot is active per FS per 16 channels.

Runs not included in the Table, show that 10 dB increase of the transmitter power has hardly any influence on blocked calls. This proves that the range is C/I limited.

The results of this simulation must not be generalized without care. At least the influence of close by adjacent systems has to be considered.

Suppose that the building is surrounded on all three floors by adjacent systems, see the squares in Fig. 3. If all systems have their frames synchronizezd, the traffic capacity decrease due to adjacent systems is estimated to be no more than 5-25 %.

If the adjacent systems are non-synchronized the capacity decrease is estimated to be 10-30 %. See Appendix. These figures have to be verified by further simulations.

Thus 4 to 8 MHz will be needed to carry 12.500 E/km^2 and 3 floors in offices. The same band can be used for residential and public telepoint applications, which have lower traffic densities than offices.

Later possible introduction of lower rate codecs will increase the capacity for speech but not for data.

Comparison with a FDMA system

The results in the Table are also applicable on a FDMA system. But in this case each fixed station, FS, will need i complete radios and a combiner. See Fig. 5. If i is chosen to give no worse than 0.5 % extra blocking per FS, than i equals the figures in the last column of the Table. Thus a FDMA system with at least 16 or 32 channels needs 3-8 times more FS radios than the TDMA system.

FDMA and TDMA portables are supposed to be equal in cost, while TDMA can give better system features and lower fixed station costs.

A comparison between this TDMA system and a FDMA system has been made by ECTEL (5).

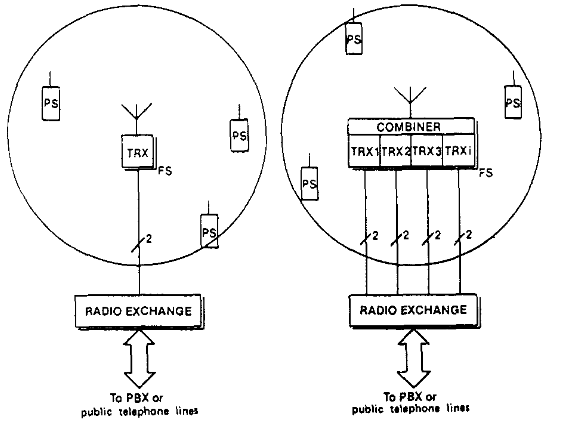


FIG. 5. SINGLE CELL TDMA (max. 16 simultaneous channels) SINGLE CELL FDMA (max. i simultaneous channels)

CONCLUSIONS

CEPT and industrial organizations in Europe are currently specifying a medium and long term application of advanced digital cordless telecommunications. The application ranges from basic cordless telephones, pico cellular office systems to Pan-European public telepoints.

Products on the market are expected in the early 1990'ies, starting with private business office systems which are regarded as most important.

The specification will allow large flexibility to continously develop private systems within the mandatory Level 1 specification (co-existence), as well as of an optional (interoperability) Level 2 specification mandatory for public telepoint speech service.

The basic channel access principle will be multicarrier TDMA/TDD, allocated around 1.6-1.7 GHz.

It is very essential to avoid traditional frequency planning and channel allocation for individual (private) systems.

The solution is Dynamic Channel Allocation, which avoids frequency planning for individual systems and permits system sales on a general (not individual) license.

From simulations and trials with a test system we have demonstrated the following:

- o TDMA/TDD is very well suited for speech and data pico cellular office systems using decentralized dynamic channel allocation and fast hand over, since one single radio can listen or communicate on all time slots simultaneously. This also gives inexpensive fixed stations (typical 4 times fewer radios than in a FDMA system) and avoids need for careful planning of the number of channels needed per fixed stations.
- o Dynamic channel allocation in combination with C/I limited range and signal strength measurements, that ensure that each portable is connected to the closest fixed station, gives a stable frequency efficient system.
- o Simple procedures ensure that the rate of interrupted calls is about 10 times below the grade of service for new calls.
- o The system can temporary assign more than one slot to a user if high data rate is needed.

- o Simulations indicate that 10 to 20 MHz can per 3 floors provide densities of 30 000 E/km^2 or over 150 000 telephones/km^2 at 0.1-1 % grade of service.
- o Switched antenna diversity performs very well, in an environment with slow movements. No time dispersion equalization is needed.
- o Slot synchronization between very close by adjacent system is desirable, but not necessary. The penalty is a graceful capacity reduction in the interference areas.
- o A modest echo control is needed on each trunk to the fixed network, but only if 2-wire inter connection is used.

APPENDIX

A.1 SLOT SYNCHRONIZATION BETWEEN SYSTEMS

Slot synchronization within the own system

Slot synchronization is easily made since all cells are connected to the same central. The accuracy is related to the size of the guard bands between the slots, and gives no problem.

Slot synchronization between systems

Fig. 6 shows that one non-synchronized slot occupies 2 slots in the adjacent cell.

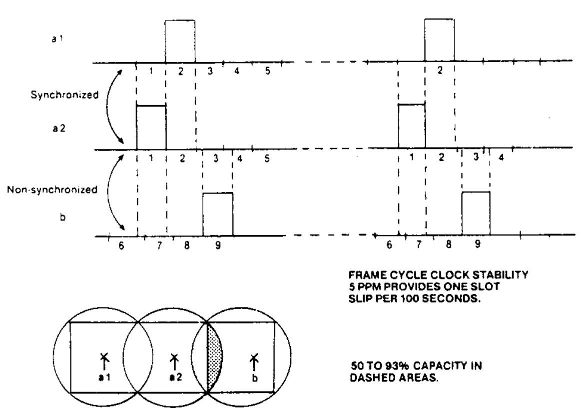


FIG. 6. SYSTEM A, WITH INTERNAL SYNCHRONIZATION, IS NOT SYNCHRONIZED WITH SYSTEM B.

A non-synchronized slot is equivalent to increased normal traffic.

2 adjacent slots will occupy 3 slots in the adjacent cell, etc. Thus a capacity decrease of 50 % will never be exceeded, not even if all base stations in the own system are mutually non-synchronized. In reality the influence will be less and only affect the interference area from the base of the neighbour system.

If the slot frame chock frequencies of the two systems are not exactly equal, than slot slips will occur with regular intervals. The shaded areas of Fig. 6 indicate slot slip areas.

The equivalent to a slot slip is that the adjacent system terminates the call on one channel (slot), and sets up a new call on another channel (slot).

However, every normal call set up can cause interference to a current call. The system with adaptive channel allocation is designed to adjust for interference from other calls. Thus the adjustment of possible interfrence from a slot slip is a normal feature, for which TDMA provides quick and elegant procedures.

It is however important that slot slips not happen to often compared to the normal call set up and termination frequency.

A frame clock stability of +-5 ppm (watch crystal) will give a slip of one slot length per 100 sec. (the length of a call). Since a base in average has 1 slot active of 16, a forced slot slip can only occur every 15th call if the base is interferred from one adjacent slot. Perhaps the same area can be interferred from 2 or 3 adjacent foreign bases, and in that case every 5th call will be caused to change channel. The density of slot slips is thus low in comparison to the rate of call set ups.

Conclusion

The conclusion (7) is that it is desirable but not necessary to have slot synchronization between adjacent systems. The penalty is a modest capacity reduction in the interference zones.

A2. ON THE NEED FOR TIME DISPERSION EQUALIZERS

Subjective conversation tests and measurements with experimental equipment (8) at sites with known RMS delay spread (up to 200 ns) show that equalizers are not needed for indoor or outdoor use at 1.3 Mbps. MSK modulation and discriminator as detector was used.

The reason is that the receivers resynchronize on each slot and that the conditions during reception are stationary (max 1 mm movement), and that corrupted bits only occur in conjunction with fading dips. For the vast majority of these fading dips the field strength is anyhow below the sensitivity level. Thus the cure is antenna diversity and not equalization.

A3. ON THE NEED FOR ECHO CONTROL

A fram cycle time of 16 ms gives an one way speech delay of 16 ms. This delay can affect the far end talker and the near end talker. The wireless telephone is supposed to be at the near end (9).

The far end talker echo

The echoes from the near end telephone comes from the hybrid and from the acoustic path. The hybrid echo is dominating over the weaker acoustic echo that is delayed 32 ms in case of a wireless connection.

The only case of importance is a satellite connection with echo control devices at the international gateways. In this case the control devices are designed to take care of the hybrid echoes, but not necessarily the delayed acoustic echoes. However, if echo supressors are used, the long hangover time (200-300 ms) in the one way speech mode also blocks the acoustic echo, provided that the telephone acoustic loss, TAL, exceeds 24 dB, so that the acoustic echo is low enough not to be detected as break in speech from the near end. In the case of echo cancellers, the acoustic echo is blocked by the non-linear processor with good margin if TAL exceeds 24 dB (10).

The near end talker echo

The wireless telephone system will need a modest echo control on trunk level to the PABX or PSTN when 2-wire connection is used (2).

For full 4-wire connection no echo control is needed. Shorter frame cycle times will decrease the echo control demand. However, echo control can only be avoided if the frame cycle time is below 1 or 2 ms. Such a short time is not practical since the capacity loss due to the increased relative portion of guard space and signalling overhead is too large.

ACKNOWLEDGEMENT

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REFERENCES

(1) H. Ochsner, "Cordless Telecommunications: Towards a Digital Standard." Eurocon '88. 8th European Conference on Electronics, Stockholm 1988.

(2) Specification 8211-A130: Technical requirements for connection of digital cordless telephones to the public switched telephone network. Swedish Telecommunications Administration, Stockholm, Sweden.

(3) ESPA. "Business Cordless Telephones." Publication No. 5.2, Sept. 1987. ESPA c/o Willems, Kapittelweg 10, NL-4827 HG Breda, The Netherlands.

(4) D. Åkerberg, B. Persson, "A digital TDMA micro cell system for business cordless telephones." Report TY 86:1076. DMR II. Rev. C. Ericsson Radio Systems AB, S-164 80 Stockholm, Sweden.

(5) ECTEL, "Report on business cordless telecommunication systems (BCT)." Doc. ECTEL/TCS 14/88. 11, Rue Hamelin, F-75783 Paris, Cedex 16, France.

(6) D. Åkerberg, "European Advances in Cordless Telephony." National Communications Forum 88, NCF 88, Chicago, USA.

(7) D. Åkerberg, "On time synchronization needs within and between TDMA systems." Report TY 87:2077. Ericsson Radio Systems AB, S-164 80 Stockholm, Sweden.

(8) D. Åkerberg, "Tests of speech quality versus RMS delay spread for a digital TDMA cordless telephone system without use of time dispersion equalizers." Report TY 87:2091. Rev. B. Ericsson Radio Systems AB, S-164 80 Stockholm, Sweden.

(9) D. Åkerberg, "Echo considerations in design of a BCT with a delay T." Report TY 86:2001 Rev. C. Ericsson Radio Systems AB, S-164 80 Stockholm, Sweden.

(10) D. Åkerberg, "On echo control when a BCT with T ms inherent one way delay is connected via a satellite link." Report TY 87:2017. Ericsson Radio Systems AB, S-164 80 Stockholm, Sweden.