

## A Radio System Proposal for Widespread Low-Power Tetherless Communications

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IEEE Transactions on Communications,  
Vol. 39, No. 2, February 1991,  
Pages 324 - 335.

# A Radio System Proposal for Widespread Low-Power Tetherless Communications

(Invited Paper)

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**Abstract**—Tetherless communications represent the fastest growing segment of the telecommunications industry. Low-power digital radio as an access technology could be integrated into a local exchange network to provide a ubiquitous personal communications network (PCN). High quality tetherless communications services that could be provided by such an exchange network based PCN are described. A possible low-power exchange access digital radio system for providing these exchange network based PCN services is discussed. The radio system uses a spectrum efficient time-division multiple-access (TDMA) architecture made possible by advanced digital signal processing techniques. Control of the frequency reuse system is described and frequency spectrum needs are indicated.

## I. INTRODUCTION

VARIOUS approaches to tetherless communications, taken together, are arguably the fastest growing segment of the telecommunications industry. These tetherless communications approaches include: a) low-power cordless telephones and their derivatives, cordless PBX and phone points,<sup>1</sup> b) radio paging, c) higher power cellular mobile telephones, including vehicular, transportable and handheld customer sets, and d) radio data systems. Each of these approaches has its own strengths, and each has significant limitations. These approaches, their strengths and limitations, and their evolutionary trends are described in [1]–[4]. The popularity of these approaches, despite their limitations, along with customers' expressed desires to overcome the limitations suggest a very large potential market for low-power, convenient, widespread tetherless portable communications. Such an approach to portable communications based on low-power exchange-access digital radio integrated with exchange network intelligence has been described in [1]–[4]. This type of widespread low-power tetherless communications has service goals similar to the goals of the "personal communicator" approaches discussed in [5]–[7]. Low-power digital radio as an access technology integrated into a local exchange network could provide a ubiquitous personal communications network (PCN).

Section II reviews some communications services that could be provided by a network-based low-power tetherless communications system. The remainder of the paper describes a possible system based on time-division multiple-access digital (TDMA) radio technology.

## II. TETHERLESS COMMUNICATIONS SERVICES

The radio part of a personal communications network for providing widespread communications to low-power voice and/

Paper approved by the Editor for Mobile Communications of the IEEE Communications Society. Manuscript received April 6, 1990; revised May 23, 1990.

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<sup>1</sup> Phone points are also referred to as phone zones and telepoints.

Low-Power Exchange Access Digital Radio Integrated With Network And Intelligence

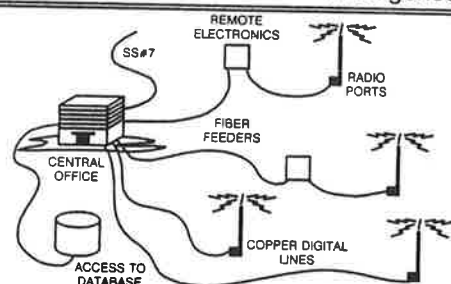


Fig. 1. Digital radio access integrated into an intelligent exchange network.

or data sets, i.e., to convenient personal communicators, requires a dense network of fixed radio ports. These radio ports need to be connected together through a copper and/or fiber distribution network that includes intelligent control logic and switching equipment. The radio ports, or minibase stations, provide radio access points to the distribution network. Such a system and network as depicted in Fig. 1 is described in [1]–[4]. In residential areas the radio port antennas would be mounted on 20–30-ft high utility poles or street light poles. Radio ports and their antennas would also be mounted inside large apartment buildings and large commercial and industrial buildings, and along streets in heavily built up urban areas. Other places where people congregate would also contain radio ports, e.g., airports, railroad stations, shopping centers, and toll plazas. Radio ports would be arranged to provide nearly continuous radio coverage throughout large service regions, e.g., entire metropolitan areas including their suburbs. The port density would also be tailored to provide enough capacity to serve the traffic density within each service area.

### A. Types of Services

While being free of a wireline tether is in itself a service of significant value, widespread tetherless communications services could be structured to be related to the amount of network facilities involved in providing them. Examples of such possible voice and/or moderate-rate data services could be the following.

1) *Neighborhood Tetherless Access*: This could be like an extended high quality cordless telephone or data service throughout a limited area around a customer's house or apartment. It would involve only a nearby central office and access to the data base containing the user's service profile and privacy encryption key, which could reside at the central office.

2) *Tetherless CENTREX*: This would be a tetherless busi-

ness voice and data service analogous to wireline voice CENTREX and would be similar to proposed wireless PBX services. Again, only a nearby central office and a data base access would be involved.

- 3) *Away-From-Home (or Office) Tetherless Access:* Such voice and data services could be distance-, and/or region-sensitive, and could invoke extra charges relative to neighborhood or CENTREX services when used in a different service area away from home or the office. The "away" access would require the network to interrogate the user's home data base and perhaps temporarily store the user's service profile and privacy encryption key in a local data base. "Away" could be indoors or outdoors in a different exchange, a different town, a different state, or even a different country or continent.
- 4) *Registration in a Home (or Office) Data base for Forwarding Voice and Data Calls to an Away Location:* This registration could be done by a user's set automatically, or under user control, when the portable set detected that it was no longer in its "regular" or "previously registered in" paging area. A data exchange "handshake" with a nearby radio port would be required, and an updating message would need to be sent by the port controller through the network to the user's home data base.
- 5) *Call Forwarding of a Page to an Away-From-Home Location in Which a User is Registered, for a Voice or Data Call Made to a User's Home Location:* The calling number could be delivered to the called person, or the caller could be informed of extra charges for a forwarded call.
- 6) *Answering a Voice or Data Call in a Call-Forwarded Area:* This service raises many questions regarding who pays for or who has the option of paying for and completing a forwarded call to a user who is registered away from his or her home area.
- 7) *Tetherless "Call for Help":* This could provide a "911"-like automatic call for help when a button is pressed a pocket carried set. The network could furnish to the emergency operator the name of the caller and the location of the port on which the call was initiated. A similar call for help could be routed by the network to a designated person or place to provide a medical alert that the subscriber was experiencing symptoms of a known medical problem.
- 8) *Other Network-Based Services:* All special network services provided to wireline users could also be provided to tetherless communication users; examples are calling number identification, selective call rejection, selective call acceptance, distinctive call ringing, voice mailbox, multiple location or multiple communication set ringing or paging, and conference calls. Many of these services have greater perceived value to a user "on the move."

The services of 3) initiating a call, 4) registering for call forwarding, 5) being paged, and 6) answering a forwarded call out of the home area are noted as possible separate services because each involves different additional network involvement in transmitting or storing user data. Since the TDMA radio link architecture described later can provide variable transmission rates in 8 kb/s increments, charges for services could be based on the total number of data and voice bits transmitted, i.e., on the product of the transmission rate and the time interval of usage.

## B. Quality of Services

Radio has a reputation for providing poor quality communication circuits in approaches like cordless and mobile telephones; but, radio provides some poor circuits only because radio systems have been statistically *designed* to include a significant number of poor quality circuits. For example, the design criterion for cellular mobile radio coverage, considering radio propagation, receiver sensitivity and cochannel interference, has been to provide good or better service in 90% of a service area during the busy hour of an average business day [8], [73]. While this is significantly better than earlier mobile system designs based on median coverage contours, it does not provide the large number of high quality circuits expected of wireline networks. It is not commensurate with the 1 or 2% call blocking criterion used in the same cellular mobile system design. That is, by design only 1 or 2 in 100 call attempts will be blocked during the busy hour, but 1 in 10 successful attempts will be judged to have a less than good quality circuit!

In contrast to mobile systems, radio transmission systems such as satellite and point-to-point digital microwave radio have stringent design criteria requiring good performance for 99.9 or even 99.99% of the time. While this often results in multipath fading margins of over 50 dB for point-to-point digital microwave radio links, and includes the penalty of battery backup power and spare satellites in satellite systems, these radio transmission circuits are usually judged to be of high quality.

The system described in the remainder of this paper is based on a coverage criterion of good or better over 99% of a service area. While this is significantly better than the 90% cellular mobile radio criterion, it may need to be made even better to provide radio access that will be subjectively judged to be as good as wireline access.

## III. SYSTEM ARCHITECTURE AND CONTROL

This section describes the architecture and control of a low-power radio system configured to provide widespread portable communications to small pocket-size voice and data sets, i.e., to "personal communicators." Parameters of the system are also given. The details of the many considerations necessary to arrive at the architecture, control, and parameters are too extensive to include in this paper. Therefore, many references to technical papers are made for the significant details needed in arriving at the system choices presented herein.

Customized digital signal processing techniques implemented in application specific integrated circuits (ASIC's) are the key to high quality digital radio links. However, digital signal processing must be used sparingly to avoid excessive power consumption. Power is a scarce commodity for pedestrian carried personal portable communications sets [1]. The power penalty for using general purpose digital signal processors is generally too great for their use in personal portable sets.

Power consumption in CMOS ASIC's is proportional to the number of operations performed per unit time and is thus proportional to computational complexity. Extensive manipulation of many high precision numbers, particularly multiplications and matrix operations, consumes significant power. Functions that require complex digital signal processing are thus to be avoided. Examples of such functions include computationally intensive forward error correction, bit rate compressed speech, and multipath delay equalization. The goal of the radio link architecture should be to minimize portable set power consumption. A long term goal should be for portable set power con-

sumption in the range of 100–200 mW or less when the set is active. This also obviously requires low transmitter power on the order of 10 mW [2], [3].

### A. Spatial Radio Port Configuration

The low transmitter power dictated by size and weight limitations on small battery-operated transceivers limits radio range to several hundred to a few thousand feet [2]–[4]. Such short range translates into a need for a dense two-dimensional arrangement of fixed radio ports in residential areas or on streets in urban areas [2]–[4]. An even more dense three-dimensional arrangement of radio ports is required inside large buildings [2] both to overcome the severe radio attenuation through walls [9]–[11], and to serve the large user densities within buildings. The arrangement of radio ports would be neither completely random nor perfectly structured. However, ports would be arranged approximately on a regular grid. Radio coverage in such a port network is highly variable because of the vagaries of propagation within and around buildings [2], [11]–[25], [70]. In fact, computer simulations of radio access in such an environment [26], [27] show that only about half of the users accessing radio ports in a typical residential environment will actually select the closest radio port. Thus, referring to average coverage area shapes as circles, squares or hexagons [28] surrounding radio ports is a largely academic exercise.

The port locations are, of course, real entities, and they can be arranged on grids in two dimensions that are approximately triangular, square, or hexagonal. These grids correspond to the often-referred-to hexagonal, square or triangular average coverage areas [28].

Because of the scarcity of radio frequencies and the large number of radio ports required, frequencies must be channelized and the frequency channels reused among radio ports having sufficient spatial separation. Factors that influence the determination of sufficient separation are a) the quality of the resulting radio channels in the presence of the fluctuating cochannel interference from other ports, b) the economics of the resulting system architecture that depends on the number of radio circuits per port, and c) the radio spectrum utilization efficiency. Trade-offs and compromises among these conflicting factors are required in specifying the port separation for frequency reuse.

The radio channel quality can be increased both by decreasing the distance between fixed radio ports to increase the signal fading margin relative to the receiver noise threshold, and by increasing the frequency reuse interval (i.e., increasing the number of radio ports between ports using the same frequency) to reduce the cochannel interference.

For a given density of users, these factors affect economics and spectrum utilization efficiency in a complex way. Decreasing the distance between ports increases the number of ports, thus increasing the cost of the fixed radio port network, but it reduces the number of radio circuits required at a port if the user density remains fixed. Thus, if the amount of spectrum remains fixed, the frequency reuse interval can be increased without affecting the overall number of customers served in a given amount of spectrum.

From another viewpoint, consider decreasing the amount of spectrum available while holding fixed the user density and quality requirement. This requires the capacity of each port to be decreased, thus requiring the distance between ports to decrease and the number of ports to increase. The cost of the port network increases accordingly. Because of the complexity of these tradeoffs and the uncertainty of the input data, a best compromise is still being sought by system architects.

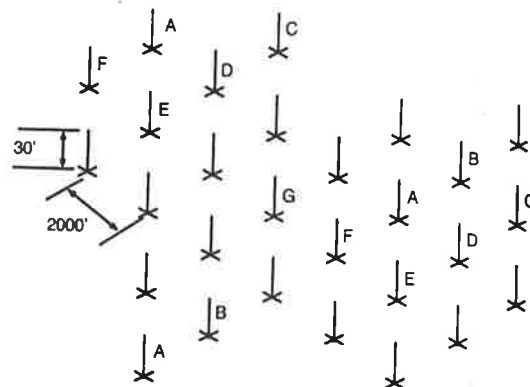


Fig. 2. Two-dimensional radio port arrangement in a residential area.

Analytical studies and computer simulations of cochannel interference statistics for triangular and square radio port grids, i.e., for idealized coverage areas based on hexagons and squares, show little difference between the geometries with the little advantage going to the square grid [29], [30]. Since square geometry is also easier to work with, the logical conceptual arrangement for radio ports is a square grid as depicted in Fig. 2. The 2000-ft port spacing and 30-ft port antenna height shown in Fig. 2 result from radio-link power-budget analyses in [2] and [3] for a residential environment.

A similar three-dimensional arrangement based on a cubic grid can be used within large multistory buildings. In large buildings, frequency channels can be reused horizontally in adjacent city blocks, and vertically between floors as discussed in [2].

### B. Modulation and Demodulation

The basic modulation compromise to be made is between more power-efficient constant-envelope modulations and more spectrum-efficient multilevel nonconstant-envelope modulations [2]. Transmitter power efficiency is not so dominant a consideration for low-power transceivers since other circuitry (e.g., modulation, demodulation, speech processing, and control) contributes significantly to power consumption. Also, new circuit techniques promise to provide linear amplifiers with acceptable dc to radio-frequency power conversion efficiency [31]. Therefore, the modulation choice has been made to provide the best spectrum-utilization efficiency.

The best modulation is then 4-QAM<sup>2</sup> with Nyquist pulse shaping. A pulse shaping factor [32] of  $\alpha = 0.5$  is a good compromise between spectrum occupancy and detectability. Either conventional 4-QAM [32] or the  $\pi/4$ -shift implementation, sometimes called  $\pi/4$ -PSK [33], yields similar performance [31].

The choice of 4-level modulation maximizes spectral efficiency considering both spectrum occupancy and resistance to cochannel interference [2], [34]. Smaller pulse shaping factors,  $\alpha$ , yield more compact spectra, but at the expense of lower resistance to cochannel interference<sup>3</sup> and to implementation imperfections.

Higher level modulation will not result in a higher allowed transmission rate for a specified bit error ratio and multipath

<sup>2</sup> This modulation is sometimes referred to as 4-PSK, but phase shift modulation is constant envelope. Nyquist filtering of quadrature signal components yields quadrature amplitude modulation which, of course, also contains  $\pi/2$  phase variations at the sampling instants.

<sup>3</sup> Lower resistance to cochannel interference results from an increased sensitivity to imperfect symbol timing; symbol timing jitter is caused by cochannel interference and noise.

delay spread [35]–[37]. While the symbol rate decreases with increasing modulation levels, this is offset nearly exactly by increased sensitivity to intersymbol interference from multipath delay spread. Note also that bandpass filter asymmetries and carrier offsets from filter center frequencies cause higher error rates for higher level modulations. These asymmetries produce effects equivalent to intersymbol interference from multipath delay spread. Therefore, higher level modulations incur greater implementation degradation for given filter tolerances; this can easily amount to a 1 or 2 dB greater departure from theoretical performance for 8- or 16-level modulations compared to 4-level. Theoretically, 8-PSK modulation with bit-dependent coding appears to have 1 or 2 dB better cochannel interference performance over 4-QAM for the same information bit rate [38]. However, this can only be realized with considerably more stringent, and thus more expensive, requirements on receiver filters and frequency stability, and with the added complexity of power consumption required for error correction decoding.

Coherent demodulation provides a 2.5–3 dB advantage against both Gaussian noise and cochannel interference compared to differential demodulation [39], [40]. Coherent demodulation also performs better against multipath delay spread [40]. Conventional implementations of coherent demodulation in TDMA radio receivers have incurred excessive symbol overhead for coherent carrier recovery. However, new digital signal-processing techniques permit coherent carrier recovery without any overhead [41], [42]. These techniques do not require multiplication, and manipulate only low precision numbers. Therefore, the advantages of coherent demodulation can be obtained with only a small increase in the complexity of a VLSI receiver circuit, and with little power consumption penalty.

### C. Transmission Rate

The transmission rate in the propagation environment within and around buildings is limited by the spread in time delays [16]–[21], [23]–[25] associated with the multiple propagation paths that result from reflections from walls and objects. Adaptive equalization of this multipath delay spread would incur significant undesirable complexity in portable radio receivers. This added complexity includes many multiplications and would incur a significant penalty in power consumption. Antenna diversity, needed to mitigate fading for areas exhibiting small delay spread, is also effective in reducing the block<sup>4</sup> error ratio produced by larger delay spreads [43]–[48], if the symbol period is larger than the delay spread, as illustrated in Fig. 3 from [43].

Multipath measurements made in 6 commercial office and laboratory buildings, in and around 3 houses, in a small city, and in several factories have consistently exhibited delay spreads of less than 0.5  $\mu$ s when outside port antenna heights were less than 30 ft and path lengths were less than 2000 ft, and when inside port and user locations were within the same building [16]–[21], [23]–[25].<sup>5</sup>

A transmission rate of 450 kb/s has been selected for the radio link described here. From Fig. 3, this should yield a block

<sup>4</sup> It is useful to consider error free groups or blocks of 100 or so bits since: a) the TDMA radio link discussed later transmits bits in groups or blocks, b) simple error detection is done on blocks of bits, c) speech is often encoded in blocks of bits, and d) data integrity is important as a group of bits representing some entity.

<sup>5</sup> These values are significantly less than the several microseconds to several tens of microseconds of delay spread found in the vehicular mobile radio environment which has base station antenna heights of hundreds of feet and paths on the order of 10 mi.

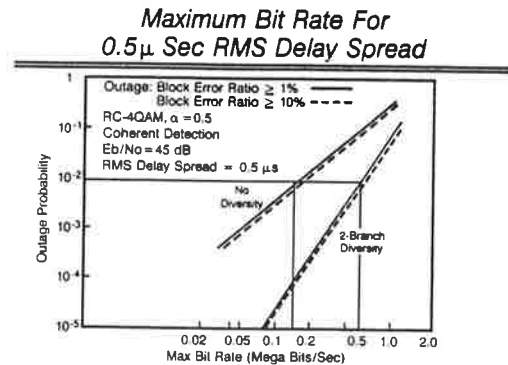


Fig. 3. Maximum bit rate for 0.5  $\mu$ s rms delay spread. Outage probability is the fraction of multipath samples that cause the block error ratio indicated in the key. The multipath samples are from a random collection that have an average rms delay spread of 0.5  $\mu$ s.

error ratio of  $10^{-2}$  or less because of delay spread, over virtually all low-power portable radio service areas for the system parameters selected. The results of Fig. 3 obtained from computer simulation also have been realized in equipment in laboratories [48]. A transmission rate of 450 kb/s, obtained without the use of multipath delay equalization, is adequate for many multiplexed voice circuits and for moderate rate data (up to a few hundred kb/s). Digital speech encoders suitable for portable communications [49] produce subjectively good speech quality at block error ratios of up to 0.01–0.04. Automatic repeating of data blocks in error (ARQ protocols) can provide good data throughput at such block error ratios.

### D. TDMA Radio Link Format

TDMA radio links have several advantages as discussed in [2, p. 459]. Briefly, a) fewer radio transceivers are required for a given number of user circuits at a radio port, reducing port complexity and cost, b) different transmission rates can be provided on demand to different users in increments of some minimum rate increment, c) nonactive portions of the TDMA frame can be used for portable sets to assess the quality of other frequency channels and to do measurements for microscopic diversity selection [47], [48], d) the diplexer function in a portable set can be a simple solid-state switch, e) frequency stability requirements are reduced, f) fewer frequency channels need to be synthesized, and g) stepping between frequency channels can be done faster. However, in the past, conventional receiver implementations have required significant bit overhead in each TDMA time slot for time slot synchronization, symbol timing and carrier recovery.

Recent new receiver implementations [41], [42], [48], based on digital signal-processing techniques using oversampling of each digital symbol, permit robust symbol timing and carrier recovery with no bit overhead, and without incurring performance degradations. These techniques manipulate low precision numbers and do not require multiplication. Therefore, they incur very little penalty in power consumption. Combining the time slot synchronization function with error detection [50]–[52] permits time slot synchronization with only 2 or 3 b of overhead per time slot.

Two-frequency TDMA radio link architecture has been selected as illustrated in Fig. 4. That is, the uplink (portable-to-port) is transmitted on one frequency while the downlink (port-to-portable) occupies a different frequency separated from the uplink by several tens of MHz. The two-frequency architecture is advantageous in residential environments having 20–30 ft high port antennas [2]. In such residential environments, the

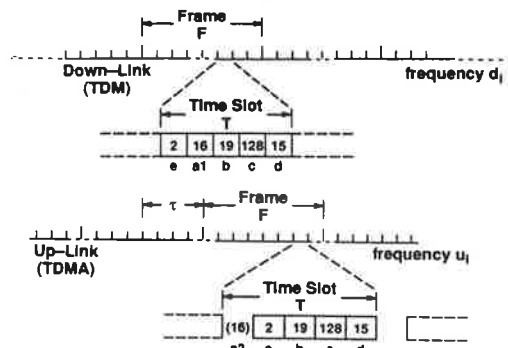


Fig. 4. Proposed TDM/TDMA frame structure.<sup>6</sup> Bit allocation is: (a2) up-link guard time, (a1) down-link synchronization, (b) system control (supervision), (c) bearer bits carrying user information, (d) redundancy bits for error detection and time slot synchronization (14 b from a (161,147) cyclic code plus one additional bit), and (e) differential encoding/decoding.  $F = 16$  ms;  $T = 0.4$  ms;  $\tau = F/2$ ; 2 b/symbol; 225 ksymbols/s.

attenuation between cochannel port antennas will often be much less than the attenuation between a cochannel port antenna and its accessing users in houses at lower heights. An alternative of single frequency TDMA (time-division duplex) in such an environment would require precise synchronization of cochannel port timing phase and relatively large guard times between uplink and downlink transmissions in order to avoid the overwhelming of users by strong port-to-port cochannel interference [2]. Two-frequency TDMA avoids the complexity of timing phase synchronization among ports. Phase synchronization among ports that are served from different central offices is a particularly difficult network obstacle to overcome.

Fig. 4 illustrates the TDM/TDMA frame structure selected for the 450 kb/s radio link [52]. The TDMA frame length is 16 ms and contains 7200 b. This selection trades the increasing transmission delay incurred with increasing frame length against the decreasing percentage overhead with increasing frame length.<sup>6</sup> The percentage overhead is the number of TDMA overhead bits divided by the total number of transmitted bits.

The 16 ms frame is divided into 40 TDMA time slots or blocks, each containing 180 b. A time slot or block is a collection of bits that are transmitted, received, and processed by the radio equipment as an entity. The bits in a time slot contain the information necessary to synchronize with the time slot, to decode the information bearing bits, to detect transmission errors, and to coordinate radio link and system control associated with the time slot. Information bearing bits in time slots separated by 16 ms, i.e., in adjacent frames, are generally collected together to form a continuous circuit. The bits in a time slot are apportioned as follows:

- 128 bearer bits carrying user information, either voice or data;
- 19 bits for continuous radio link and system control (supervision);
- 14 redundancy bits for a (161,147) cyclic code used for both error detection and time slot synchronization [50]–[52]. Only a 3 bit penalty is paid from the error detection effectiveness for the synchronization function, i.e., the error detection effectiveness is reduced to that of 11 bits of redundancy. However, 3 bits are significantly fewer than would be needed to achieve the same synchro-

nization effectiveness with a separate synchronization sequence [53] (e.g., a Barker code);

- 16 bits for guard time in the uplink direction (portable port); these bits are used in the downlink direction for frame synchronization pattern in each time slot;
- 2 leading bits for differential encoding/decoding of coherently detected bits; this is needed to remove the ambiguity in the carrier phase recovery;
- 1 additional bit to make the number of symbols in a slot an integer; this extra bit can provide additional error detection redundancy.

The functions of the 19 control bits are not presently defined; however, 3 or 4 will be used to provide discrimination among adjacent cochannel ports. This number could possibly be reduced to 8 control bits with the use of a (151,136) cyclic code. The control information is spread over several frames.

The 16 b (8 symbols) uplink guard time includes the following allocations: a) 3 symbols (6 b) for roundtrip propagation delay of about three interport spacings of 2000 ft each; no attempt is made to compensate for this delay since it is so small for short paths involved; b) 2 symbols (4 b) for timing tolerance between port and portable synchronization; and c) 3 symbols (6 b) for turning the portable transmitter on or off. Transmitter turn over distances greater than one port spacing is included because the access process in the highly variable propagation environment frequently results in portables accessing ports more than one interport spacing away [26]. A guard time of 3 symbols transmitter turn on and off insures 60 dB or greater adjacent frequency channel isolation [79].

The 128 information bearing bits in a time slot result in an information rate of 8 kb/s for a circuit. On demand when a radio circuit is set up, two of these time slots can be combined to support 16 kb/s, four can be combined to support 32 kb/s, etc. Thus, this flexible radio link architecture can provide various transmission rate circuits on demand, in increments of 8 kb/s to 320 kb/s. It is possible to form lower rate circuits combining bits from every  $n$ th frame where the likely range  $n$  could be from 2 to 6 or 8.

Note that there are no overhead bits allocated in the TDMA time slot for either symbol timing or carrier recovery. The functions are implemented in novel digital signal processing techniques that use all of the bits in the time slot [41], [42], [43], [52]. The current implementation incurs a processing delay of 0.8 ms. This could be reduced to 1 time slot (0.4 ms) or less with higher-speed digital signal processing circuitry. The only TDMA overhead bits that would not be incurred in another digital radio approach, e.g., single channel per carrier (SCPC) frequency division multiple access (FDMA), are the 16 b of guard time plus the 2 differential decoding bits. Thus, the total TDMA overhead is 18 b out of 180 or 10%.

### E. Antenna Diversity

Antenna diversity is a very beneficial technique [72] for use in low-power hand-held portable communications. In this environment, antenna diversity reduces signal fluctuations caused by multipath propagation (microscopic diversity) and by random orientation of portable-set antennas [54], [55]. For a given signal quality requirement, this reduction in signal fluctuation permits either a decrease of transmitter power or an increase in range. It also permits a significant decrease in frequency-reuse interval [26], [30], [56], [57]. Antenna diversity also permits an increase in transmission rate because it reduces the digital error ratio  $\delta$

<sup>6</sup> An 8 ms frame length with suitably adjusted fewer redundancy and control bits would also be an acceptable compromise. Since this paper was written, a new low-delay TDMA frame structure has been proposed and is being implemented in the laboratory [80].

a given delay spread when the symbol period is larger than the multipath delay spread [43]–[48].

Measurements have been made within and around buildings of the crosspolarization coupling magnitude [12], [58] and the decorrelation of crosspolarized and copolarized signals [59]. Diversity is needed only in areas where signal attenuation is high. The measurements show that the crosspolarization is strong enough and the crosspolarized signals are decorrelated enough in high attenuation areas to make diversity very effective using colocated antennas having different polarizations [2], [54], [55]. Two colocated loop antennas placed within a portable set would realize this antenna characteristic.

In conventional diversity receiver implementations, the realization of effective diversity has required two receivers, thus increasing set complexity. However, in the slowly varying multipath environment of portable radio, a TDMA radio link architecture permits the realization of fully effective selection diversity in a portable set using only one receiver [47], [48], [60]. If the port transmitter is always turned on during the TDMA time slot preceding the time slot or slots in which data bits are transmitted to the portable set, the portable set can switch between antennas during that preceding time slot, measure which antenna is receiving the best signal, and use the best signal antenna for receiving its data time slot. Laboratory implementations and measurements demonstrating the effectiveness of this inexpensive diversity approach are reported in [47], [48], [60].

#### F. Channel<sup>7</sup> Coding / Speech Coding

It is well known that error-correction channel<sup>7</sup> coding reduces the signal-to-noise and signal-to-interference ratio required for a given bit error ratio as long as the errors occur randomly. However, this improvement is realized by adding redundant bits and thus expanding the bandwidth required. It also incurs substantial computational complexity, and this increases power consumption. Also, the slow signal fading resulting from users sitting or walking causes portable radio multipath channels to have very bursty error patterns. During long periods of strong signal no errors occur, so error correction is not needed. Conversely, even with diversity there are intervals of low signal that have so many errors that error correction is ineffective. Quantitative studies show that, in slow-fading portable radio environments with bursty errors, the improvement in signal-to-interference ratio from error correction [61], [72] is not adequate to offset the bandwidth expansion penalty incurred. The additional errors that occur because of delay spread [43], [44], and because of imperfect synchronization [62], [74] in fading channels further increase the burstiness of the error patterns and thus further decrease the effectiveness of error correction. Bit interleaving used to randomize bursty error patterns in fast fading channels is ineffective for slowly fading portable radio channels because the interleaving delay required to randomize the errors would cause excessive degradation of the perceptual quality of the channels for two-way speech.

Error-detection channel coding requires significantly fewer redundant bits than error correction, to be effective against any given number of bits in error. Computation required for error detection is also relatively simple and thus incurs little power consumption penalty. Error detection can be used to detect TDMA time slots containing errors, and digitally encoded speech

can be extrapolated over the errored time slots using data from previous time slots [49]. Extrapolation can improve the perceptual quality of speech enough to make simple error detection competitive with error correction in the portable radio environment. Both natural redundancy in speech and the removal of noise peaks caused by errors in most-significant-bits contribute to perceptual improvement. Of course, error detection also can be used to request data repeats for data transmission errors. This incurs delay and throughput reduction that, while unacceptable for voice transmission, is preferred over accepting errors in data transmission.

The use of error detection is made even more spectrally efficient when a cyclic code and a marker are also used to provide TDMA time slot synchronization [50]–[52], as noted in Section III-D. Experimental laboratory equipment has demonstrated the robustness of this combined error detection/synchronization technique to multipath fading and transmitter/receiver frequency offset [48], [52].

Several techniques for digitally encoding speech at bit rates of 8 to 16 kb/s are being actively researched throughout the world [63]. It is beyond the scope of this paper to treat this subject in detail. The intense computational complexity of these approaches, that includes many multiplications, incurs excessive power consumption and processing delay for their current use in convenient pocket-carried equipment. However, rapid advances in low-power CMOS very-large-scale integrated (VLSI) circuits and in speech coding algorithm implementations may make these encoding rates practical soon. The flexible TDMA radio-link format can easily accommodate less complex 32 kb/s speech encoding for initial deployment, and also can accommodate 16 and 8 kb/s encoding when they become practical. The network interfaces required for different speech-encoding algorithms can be implemented with different software in digital signal processors. The speech coding algorithm implemented in a portable can be provided to the network interface when the call is set up, either via transmission from the portable or from the user profile stored in the network database.

#### G. System Control

The access strategies or algorithms used by a portable radio to choose the frequency channel and radio port for setting up a circuit are probably the least well understood facet of frequency-reusing portable radio systems. The choice of channel access strategy has a significant impact on the performance of the system and on the frequency-reuse interval required [56], [57]. The strategy includes the signal parameters to be measured, the selection algorithm to be implemented, and the arrangement of frequency channels and time slots in space. Since many radio ports, and even more portable sets, would be required to field-test different access strategies, such direct testing is not practical. The only practical way to compare the performance of different access strategies is to large-scale computer simulations [30], [56], [57], [61], [67] that incorporate radio propagation models [12]–[15], [20]–[22], [71] obtained from extensive field measurements.

Earlier simulations showed the effectiveness of macroscopic diversity among cooperating radio ports for mitigating the large-scale (macroscopic) signal variations that result from shadowing by walls and terrain features [30], [61]. Propagation measurements have confirmed this effectiveness [71]. Macroscopic diversity can be realized if the access strategy [56], [57] is continuously repeated by the portable set during the progress of a call, and if the radio circuit is transferred to another port

<sup>7</sup> In this section the convention is followed of referring to error control coding on a radio channel as channel coding; however, as used in this TDMA application, coding is applied separately to each time slot in each frame.

when that port becomes a better choice. That is, transfer to a better port is made before the circuit to the previous port degrades significantly. To implement a continuously repeating access strategy, a portable set must make signal measurements during time slots in the TDMA frame within which it is not transmitting or receiving bits needed to complete its transmission circuit. This requires a port to transmit in a few time slots more than are needed to provide the circuit required by a given portable set. Results from many simulations have assumed continuous transmission from all ports in a time division multiplex (TDM) format [56], [57], [64]. An idle-circuit bit-pattern would be inserted into time slots not in use. TDM port transmissions result in a worst case interference environment for the radio downlinks (port-to-portables), since all interfering ports are active in all time slots.

With TDM downlinks, computer simulations show that measuring signal-to-interference ratio ( $S/I$ ) at the portable, and choosing the frequency channel with the highest  $S/I$ , results in the best  $S/I$  statistics after channel access [56], [57]. The portable-to-port transmission direction (uplink) is TDMA, with only the time slots providing active circuits being occupied by portable set transmissions. The uplink interference environment is more complex than the downlink because portable sets are randomly located in space, and their transmission frequencies result from the downlink channel access procedure. Other factors discussed in [64], [65] also contribute to this interference environment imbalance. Simulations show that the overall  $S/I$  performance of the TDMA uplinks can be improved by implementing time slot management strategies [68] that save the "best" time slots for the portables that have the lowest radio signals at the port. The best time slots are those containing the least interference. For propagation parameters consistent with field measurements, and for traffic intensities producing blocking of 1–2%, these time-slot-management strategies can yield uplink  $S/I$  values that are somewhat better (3–5 dB) than TDM downlink  $S/I$  values for 99% of user.

The downlink  $S/I$  statistics could be improved by turning off time slots not needed either for transmitting to users or for providing diversity measurement signals for them, i.e., by using TDMA on the downlink also. Since time slot occupancy is between 0.5 and 0.6 for access attempt blocking of 1–2% for 20 circuits,<sup>8</sup> the average interference level for TDMA downlinks would be about 3 dB less than for TDM downlinks.<sup>9</sup> Macroscopic diversity implemented by monitoring signal quality on other frequency channels and transferring active circuits to "better" channels can be implemented with TDMA downlinks. This requires that all ports transmit in a few more time slots than are needed to provide the circuits required by each portable set. The apparent downlink  $S/I$  improvement for TDMA downlinks would be bought at the expense of more complex and less effective transfer of active calls among ports for providing macroscopic diversity because continuous downlink signals from other ports would not be available on which measurements could be made continuously during all nonactive time slot intervals. Therefore, it is not clear that overall system performance would significantly improve for TDMA downlinks.

<sup>8</sup> This is based on an Erlang B blocked calls cleared queueing discipline [66].

<sup>9</sup> At the point on the statistical distribution where 99% of users have a better  $S/I$ , uplink TDMA time slot management strategies [68] provide significantly better (8–10 dB)  $S/I$  than random time slot assignment. Thus, the advantage of TDMA downlinks over TDM may be significantly greater at the 99% point than indicated by the 3 dB improvement at the average.

In order to provide continuous positive system control in the time-varying signal and interference environment, radio ports must always transmit on at least one time slot, even when they are not serving any users.

Quasi-fixed frequency channel assignment to radio ports, as illustrated by letters designating different frequency channels in Fig. 2, provides good port-trunking efficiency in all environments. This permits port costs to be shared among many users. One method by which frequency channels could be allocated to ports is by having ports measure their interference environments and independently iterate their choices of channels to use [67]. Computer simulation shows that this iterative measurement/assignment procedure converges rapidly to a choice that yields good  $S/I$  performance statistics. The procedure could be implemented during low-traffic periods, e.g., 3 a.m., after new ports are added to a system.

Synchronization of time-slot-timing phase at all ports throughout a widespread portable communications system extending over regions controlled from different central offices would place a large burden on the infrastructure network. A penalty for not synchronizing time phase can occur because nonalignment of time slots can cause each uplink time slot to be exposed to two separate collections of randomly varying interferences. Computer simulation shows that the uplink  $S/I$  penalty can approach 3–6 dB at the 99th percentile on  $S/I$  distributions [68]. This tends to offset the TDM penalty on the downlinks.

Frequency reuse requirements are extremely sensitive to propagation model parameters [27]. Unfortunately, these parameters vary widely over the range of environments in which portable communications must be provided [11]–[25]. Adequate frequency spectrum must be allocated to serve a wide range of environments, not only the environments that yield the lowest spectrum requirements. Quantitative examples are given in Section III-H2).

One possible access and control sequence for operating the TDM/TDMA frequency reuse system is outlined below. For this sequence, an idle time slot on each TDM radio link is marked as a paging slot and contains the identification of the port.<sup>10</sup> The port identification includes the identity of the paging area. The size of the paging area will be determined by customer density, paging capacity and other constraints, e.g., whether the port is within a radio CENTREX boundary. If all slots are occupied, no paging slot will be available at that port at that time, and portables will need to monitor adjacent-port paging time slots. This is acceptable because, if all slots are occupied, a radio circuit cannot be set up to the port anyway.

The access and control sequences is as follows:

#### 1) Portable Set Turn-On:

- Portable set tunes through all system frequency channels one at a time and measures both relative channel power level and signal quality. The quality measurement is based on fluctuation of the digital "eye opening" and includes the impairment effects of noise- and delay-spread-produced intersymbol interference, as well as cochannel and adjacent channel interference [41], [42]. The quality measurement is available as an output from the symbol-timing procedure, and is accurate enough over the signal-to-impairment ratio ( $S/G$ ) range of 5–20 dB to provide reliable system control.
- Portable tunes to the frequency channel with largest  $S/G$ ,

<sup>10</sup> Alternative paging approaches using conventional paging systems could also be implemented.

or if the  $S/G$  is greater than 20 dB, to the channel with greatest power. Portable synchronizes [52] and scans time slots for the marked paging slot. If no paging slot is marked, the portable set tunes to the frequency channel with the second largest  $S/G$ , etc. If the  $S/G$  of the best channel or next channel in sequence is below an acceptable threshold (perhaps 15–20 dB), the portable set displays a no-service indication; otherwise it continues through the sequence.

- Portable set detects the paging area identification and compares it with the area identification it has stored as the last area in which it has registered.
- If the paging area is the same as the last area registered in, the portable set goes to "listening-for-page" mode [3] below].

**2) Portable Set Registration in a New Paging Area:** If the portable determines that it is not in the paging area in which it last registered, either because the set was just turned on in a different area or because the set was moved out of range of the port it was monitoring for a page, one of the following user-option actions will occur.

*a) User Controlled Registration:*

- Portable notifies user (beeps, vibrates, etc.) that it is in a new paging area. User can ignore notification, or command registration sequence in b) below.

*b) Automatic Registration:*

- Portable transmitter turns on and transmits a registration message addressed to the port identification number of the selected port. The message includes the portable-set identification number.
- Portable listens on the marked paging time slot for acknowledgment of registration message; if acknowledgment is not received, the portable tries again. Registration attempts are repeated several times with random delay (between 5 and 10 times). If some specified number of attempts is exceeded, the portable tunes to next best frequency channel (port), in sequence, that has  $S/G$  greater than the threshold value and tries again to register. If either the number of frequency channels (ports) attempted is greater than 4 or 5, or the next frequency to be attempted is below the  $S/G$  threshold value, a "no service" message is displayed.
- On receiving and acknowledging a registration message from a portable, the port controller forwards the message to the home database of the user along with the identification of its new paging area. This data base update is best done using out-of-band signaling such as CCS 7 using the CCITT #7, i.e., SS #7, signaling protocol [69], [77]. In some networks, a local database for the region around the port may also store the portable set identification to expedite calls to that portable from the region in which it is now located. Here the user service profile and encryption information must be sent from the home database when it receives the location registration update.

**3) Listening for Page:**

- Portable retains synchronization with the paging time slot and listens for a page of its identification. In order to save battery power, the paging protocols may be configured so

that pages are only made periodically at times also synchronized between the port and listening portables. This procedure would permit most of the portable receiver to be powered down to save battery drain during times when pages are not being sent. A shut-down period of a second or two would not cause excessive delay in the calling process, but would significantly extend portable battery time in the listen-for-page mode.

- Portable set would continue to listen on the frequency channel and time slot selected unless received signal quality deteriorates below the level that provides reliable detection of paging messages. If the received signal deteriorates below the acceptable threshold, the portable would restart a frequency channel search sequence as at 1) above.

**4) Radio Link Setup for a Call:** The portable will set up a radio link (on one or more time slots) either because it receives a page and the user answers, or because the user begins a call. A page will be forwarded to the user over the paging slot at her/his last registered location by network logic and signaling when it obtains the location from either the home database or a temporary database near the user if she/he is away from home.

- Portable checks all frequency channels as described in 1) above and determines the best frequency channel available for radio link setup.
- Portable sends a link set-up message to the port. Either the uplink time slot associated with the paging time slot or another marked idle slot could be used for this initial signaling. The message includes a) the identification number of the port being addressed, b) the user identification number, c) the number of time slots needed, and d) identification of whether the link will be for voice and the speech-coding algorithm supported, or will be for data and the data rate, whether it is packet for a contention time slot, or stream data, etc. Portable listens for acknowledgment and, if needed, follows a contention resolution and channel search procedure similar to that discussed under registration in 2) above.
- On receiving a link set-up message, a port controller retrieves the user's service profile and encryption key<sup>11</sup> from the appropriate network database via an out-of-band signaling network.
- The port and portable exchange short messages to agree on time-slot assignments and synchronize the link between them on the agreed time slot.
- The portable and port exchange encryption-synchronization messages and start the link. Note that user authentication is automatic in that if a user's set does not have the same encryption key that the network is using, link setup will fail. The encryption key is never passed over a radio link. It is only passed over the wireline signaling network, and separately stored in the user's set; therefore, radio link privacy and security is equivalent to that of the wireline network for an encryption key of adequate length.

**5) Radio Link Quality Maintenance:** The interference environment is continuously changing because new radio links will be set up and existing links will cease operation. Also, some users will move around, causing signal and interference levels to change. Thus, constant monitoring of signal and interference conditions is required. Transfers of radio links from one time

<sup>11</sup> Public key encryption [78] could be used, however, no advantage has been identified since a database access is needed anyway for user verification.

slot to another and from one radio port to another will be required to maintain the quality of some links.

- Radio ports will continuously measure  $S/G$  and power level in each time slot. From these measurements, estimates can be made of received signal level for each radio link and of impairment level for each time slot. Time slot reassignment will be initiated if  $S/G$  becomes either too small for adequate link margin or so large that the margin is excessive. The same time-slot quality criterion as used in time-slot management at initial radio-link setup also will be applied for reassignment. The port will send a time-slot change message using some of the bits in the time slot that are reserved for system control. The portable will acknowledge the message and agree when the change will occur, e.g., in the  $n$ th frame after the port acknowledges the portable's acknowledgment.
- Portable sets will rapidly tune to other frequency channels during time slots for which they are not receiving or transmitting their information or measuring signal quality for diversity antenna selection. Channel power level and signal quality will be measured in the other channels as described in step 1) above. Note, for a TDM downlink and a 32 kb/s radio link (i.e., 4 of the 8 kb/s time slots), a portable could tune over 14 frequency channels during one frame if required. This allows one diversity measurement time slot for each of the 4 information transmitting slots, and assumes the duration of one slot is required to tune between channels and stabilize. Thus, for example, 70 channels could be measured every 5 frames, i.e., every 80 ms for 16 ms frames.<sup>12</sup> However, power will be consumed by the tuning and measuring procedure, and large-scale changes are likely to occur on a scale of seconds for pedestrian users. Therefore, scanning an entire set of channels every one half to one second should be adequate.
- Radio link transfer to another port will be initiated if another frequency channel becomes significantly better than the existing channel. Before starting a transfer, the portable will determine if an adequate number of time slots is idle at the candidate port by synchronizing with the candidate port and scanning an entire frame. During this scan, the portable may lose one frame of information from the existing radio link. This lost frame will be treated like a frame lost because of propagation conditions or impulse noise, i.e., extrapolation from the previous frame for voice [49] or a repeat request for data. The request for link transfer to the candidate port could be made either to the candidate port via all the information bits in the candidate time slots or to the existing link port via the control bits in the active time slot. If candidate time slots at the candidate port are not coincident in time with slots being used for the existing link, the radio link to the new port can be established before the old link is severed. This will be the case most of the time with the average occupancies of less than 0.6 that result from access attempt blocking of 1% or less. In the few cases where noncoincident slots do not exist, the link transfer will require starting the new link in the frame immediately following the frame in which the old link is

abandoned. The old link should only be completely severed after the new link has been successfully established.

#### H. Frequency Spectrum Needs

1) *Operating Frequency:* A frequency allocation is not currently available (in 1990) for widespread low-power tetherless communications services. This issue is discussed in [1], [2]. The recently introduced bill HR 2965 in the U.S. Congress, the recent U.S. FCC NOI docket 90-314 on Personal Communications Service (PCS), the recent initiative by the British Department of Trade and Industry for Personal Communications Networks (PCN), the activity in Europe toward a third-generation Digital European Cordless Telephone (DECT), and the 1992 World Administrative Radio Conference (WARC) may provide frequencies for these services. Technical considerations [2] suggest frequencies between 0.5 and 5 GHz would be suitable for such a system, with frequencies between 1 and 3 GHz being preferred.

2) *Total Bandwidth:* The bandwidth needed for widespread low-power tetherless communications is a complex compromise involving a) the quality of the radio channels in the presence of fluctuating cochannel interference, b) the economics of the system architecture that depends on the number of radio circuits per port and c) the radio-spectrum utilization efficiency. These factors are discussed in Sections II-B and III-A. An interim working party (IWP 8/13) of the International Consultative Committee on Radio (CCIR) has recommended that a total bandwidth of 60 MHz is needed for these low-power services [75]. This section considers bandwidth needs of a single service provider for two common environments. The large multistory commercial building environment is at the high user-density extreme where bandwidth is needed to serve the large concentration of customers. Some large buildings would likely be served by one service provider. The residential environment with separated single-family houses presents a low user density where economics dictate sharing a radio port and associated distribution wire (or fiber) with many customers, and where all residences may not have portable radio communications initially.

a) *Residential Example:* Consider first a residential example with single-family houses. Adjacent channel interference can be maintained adequately low with three one-way TDM or TDMA frequency channels per MHz of bandwidth using the radio link parameters described in this paper [31]. Thus, a two-way, two-frequency<sup>13</sup> TDM/TDMA link will require 2/3 MHz. Computer simulations [26], [27], [30], [56], [57], [64], [65] consistently show a need for dividing the spectrum into 25 sets of channels for frequency reuse in two dimensions. This is based on the criterion of providing 99% good circuits in the cochannel interference environment (see Section II-B), and on two-way transmission with a range of propagation parameters representative of measurements in the residential environment [12], [13].

A housing density of 6 houses/acre, i.e., about 1500 houses/km<sup>2</sup> is representative of urban or dense suburban areas. With a port spacing of 2000 ft (600M) this yields about 550 houses/port. For many years after the start of deployment, not all houses will be users of such a service. Consider for such dense neighborhoods an initial market penetration<sup>14</sup> of 25% or a

<sup>12</sup> If TDMA downlinks were used, a portable would need to listen for a significant part of a 16 ms frame to be sure to have received an active time slot from the strongest port on a given frequency. This would significantly increase the time to scan all channels. For example, if only 2 channels could be measured each frame, it could take over 1/2 s to measure 70 channels.

<sup>13</sup> See [2, p. 459] for why single-frequency TDMA, also referred to as time division duplex (TDD), is not desirable in residential areas with port antennas on poles 20–30 ft (10 m) high.

<sup>14</sup> Some projections suggest that 50% of the adult population will have tetherless communications by the year 2000.

user density of about 140 users/port. The number of users per circuit for a specified access attempt blocking can be determined from telephone traffic (queueing) theory [66], if the statistics of the user traffic are known and are stable. Small groups of users have large statistical fluctuations in traffic. Thus, for a blocking of 1% and a typical offered traffic of 0.06 Erlangs/user, 16 or 17 circuits or about 8 users/circuit would be acceptable for serving 140 users. However, in actual concentration of traffic in the telephone plant [76], lower concentration ratios are used because of statistical demand fluctuations and even lower blocking objectives. For example, 96 users are concentrated into 24 circuits [76] where, for 0.1% blocking, a need for only 15 circuits is calculated for 96 users at 0.06 Erlang/user. That is, acceptable concentration for 96 users is 4 users/circuit where 6.4 users/circuit would be expected from a calculation based on average statistics. Adjusting the calculated 8 users/circuit by the same factor (4/6.4) yields a concentration factor of 5 users/circuit for 140 users, or a need for 28 circuits/port.

As discussed in Section III-F, power consumption and voice quality limitations in pocket-carried portable sets will likely dictate 32 kb/s speech coding for initial implementations<sup>15</sup> and for the near future. The TDMA radio link frequency channel described herein will support 10 circuits/channel at 32 kb/s/circuit. Thus, 3 TDMA radio channels per port are needed to provide the 28 circuits/port.

Combining 3 channels/port with the 25 channel sets and 2/3 MHz/channel yields a total spectrum bandwidth need of 50 MHz, which is consistent with the 60 MHz recommendation of the CCIR IWP.

For more dense housing or a larger market demand, the ports would have to be spaced closer than 2000 ft. For significantly lower housing density, ports would be equipped with fewer transceivers.

Future traffic growth beyond this example is expected in a) penetration beyond 0.25 users/house and b) additional demand for data services. Such increased demand could cause saturation of capacity, but future improvements in speech-coding implementation are expected to permit high-quality low-delay coding at 16 kb/s or lower with power consumption low enough for use in pocket-carried sets [63]. Thus, as demand increases, technology will permit the needed capacity expansion within the flexible TDMA radio link format. Note that as market penetration approaches 50%, the 50 MHz bandwidth will still be adequate with 16 kb/s speech coding.

In regions where large multistory apartment buildings are located among dense single-family residences, one set or a partial set of 25 TDMA channels can be assigned to ports within an apartment building or buildings. The remainder of the channels can serve the surrounding houses with a somewhat closer port spacing, if needed to serve the traffic density.

*b) Multistory Commercial Building Example:* The most user-dense environment exists within large multistory buildings in heavily built-up areas of cities. Some of these buildings contain large open areas comprising almost an entire floor with the floor densely occupied with desks, file cabinets, and/or low bookshelves. A typical building [20] is 200 ft  $\times$  200 ft (about 60M  $\times$  60M) with 100 ft<sup>2</sup> (about 9.3M<sup>2</sup>) allocated to each

person, yielding a density of 400 people/floor. Because of the large open area, propagation approximates inverse-distance-squared [24], and frequency reuse within the 200 ft  $\times$  200 ft area will not be possible.

Business traffic offered per user is at least a factor of 2 greater than residential traffic, so concentration is less by almost a factor of 2. For a concentration factor of 3, a need for 133 circuits per floor results. Propagation between floors is highly variable [15], [22], and detailed computer simulations of this environment are only beginning [81]. However, a reuse interval of every third floor (see [2, p. 443] for 3-D reuse) should be possible in almost all buildings. A few special-case buildings may require special treatment (e.g., placement of metal foil in ceilings or floors, distributed antennas, or leaky cables) to make this possible. The resulting circuit requirement is then 400 circuits/building, with the frequencies providing these 400 circuits repeated every three floors vertically. Frequencies can usually be reused between buildings [2, p. 443]. Using the 10 circuits/TDMA channel and 2/3 MHz/TDMA channel from the previous section results in a need for about 26.7 MHz/building for voice alone. Consider also that TDMA channels will be needed outside on the sidewalks and streets, and these channels must be separate from those used inside buildings (see [2, p. 444]). With the two-dimensional reuse of 25 channel sets from the previous section used for the streets, an additional 16.7 MHz is needed, again for voice alone. It is virtually impossible to determine moderate-rate radio data needs in this environment at this time. However, current moderate-rate data needs, although bursty, require less total bandwidth than voice. If one makes a "best guess" and adds 15% more capacity initially to accommodate some data, the total 43.4 MHz voice need becomes about 50 MHz and is comparable to the need in residential areas. The expected increasing demand for data can be accommodated in the future by the decreasing voice bandwidth requirement of a transition to 16 kb/s or lower speech coding.

#### IV. SUMMARY AND CONCLUSION

The rapid growth in many limited approaches to tetherless communications suggests a very large user need and potential world-wide market for more-universal widespread low-power personal portable communications. Convenient economical communications to low-power pocket-size "personal communicators" could be provided by a widespread personal communications network (PCN) based on low-power digital-radio access to the intelligent local exchange network [1]. A TDMA radio link architecture and intelligent control of radio frequency channel and time slot assignment can provide low cost, high radio-circuit quality, and good spectrum efficiency. This is made possible by new sophisticated low-complexity digital signal-processing techniques implemented in very-large-scale integrated circuits (VLSI). The exchange-network-based PCN described in this paper could make exciting and useful new tetherless communications services available to everyone, within a modest 50 MHz frequency spectrum requirement per service provider, when and if the electropolitical and regulatory obstacles can be overcome.

#### V. ACKNOWLEDGMENT

Many have contributed to the tetherless personal portable communications system proposed in this paper. The major contributors are represented in the references to their work that form a basis for the system architecture and parameters. The contributions of H. W. Arnold and P. T. Porter to the overall

<sup>15</sup> This is consistent with the low-power pocket phone development activity in Europe, e.g., the CT-2 phone point in the U.K. and the Digital European Cordless Telephone (DECT) being standardized by the European Telecommunications Standards Institute (ETSI). Conversely, vehicular systems that have fewer constraints on power consumption and voice quality can consider speech coding rates as low as 8 kb/s now.

applied research underlying this system are especially significant. I thank them for their outstanding contributions.

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## A Layered Network Protocol for Packet Voice and Data Integration

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IEEE Journal on Selected Areas in Communications,  
Vol. SAC-1, No. 6, December 1983,  
Pages 1006 - 1013.

# A Layered Network Protocol for Packet Voice and Data Integration

WILLIAM L. HOBERECHT

**Abstract**—This paper discusses considerations in the design of packet protocols suitable for interactive voice and interactive data communication, and then outlines a potential layered protocol architecture for the internal communication of a long haul network that might support packet voice and packet data transport.

Following the protocol description, the paper compares potential delays for two voice/data packet network architectures: one using only link retransmission, the other using only edge retransmission for data (as included by the voice/data protocol). The underlying data traffic loads offered to the network are the same for the two methods, although they give rise to different traffic patterns. This preliminary analysis shows that the average delay using an edge-to-edge recovery discipline can be made comparable to the delays introduced with a link-by-link recovery discipline, if the network uses high speed transmission facilities (e.g., over 1 Mbits/s) having good error characteristics (e.g., one or less packets corrupted in each 1000), and sends up to 128 bytes of customer data as a single packet.

## I. INTRODUCTION

THE similarity of packet voice and packet data transport, recognized by many [1]–[4], suggests further study in the area of designing an integrated voice/data network. Unfortunately, along with these similarities are several conflicting communication requirements. Data applications demand essentially error-free transmission and existing protocol standards (e.g., CCITT Recommendation X.25) reflect this need. On the other hand, voice is tolerant of occasional errors. The inherent characteristics of voice allow small amounts of lost or corrupted information to be reconstructed, or even omitted, without a severe degradation in voice quality. Voice, however does have more stringent requirements on the amount of delay permissible between the time an utterance is spoken and the time it is subsequently heard by the receiver. While the exact limits on the amount of acceptable delay are unknown, subjective tests have shown that excessive delays may not be tolerated by telephone users when nondelayed phone service is available [5], [6] (cost of service was not a parameter in these tests).

One recent study [7] addressed the issue of voice/data packet protocols and described a point-to-point protocol meeting the needs of both voice and data. Some earlier studies in the area of protocols for packetized voice [8], [9] have concentrated on end-to-end aspects of the protocol

(i.e., peer communication between packet voice transmitters and receivers) and on using current packet data networks for packet voice transport. These efforts included a demonstration of packet voice over Arpanet, reported in 1978 [10], suggesting that it may also be possible to use existing packet data networks (or at least their protocol architecture) for packet voice. While this earlier work considered use of current network designs for packet voice transmission, this paper presents an alternative view of a layered protocol structure for the *internal* operation of a long haul packet network that can support packet voice and packet data transport. These discussions and the description of the protocol architecture represent preliminary thinking on concepts that might be incorporated into an integrated packet voice/data system. The intent is to stimulate further thought on possible protocol architectures for voice/data packet transport.

The remainder of this paper describes packet voice communication and a possible internal network packet voice/data protocol structure. The paper first presents some of the factors that have a pronounced effect on the protocol design and the placement of protocol functions, and then describes a layered protocol for the internal communication of a voice/data packet network. The final section presents an analysis of delay for data using the voice/data packet protocol and compares it to delay encountered using a traditional error recovery protocol.

Throughout the paper, a distinction is made between edge-to-edge protocols (completely contained within the network, exercised between packet network interfaces) and end-to-end protocols (exercised between network customers).

## II. PACKET VOICE/PACKET DATA NETWORK COMPONENTS

Fig. 1 shows the components of a generic packet network:

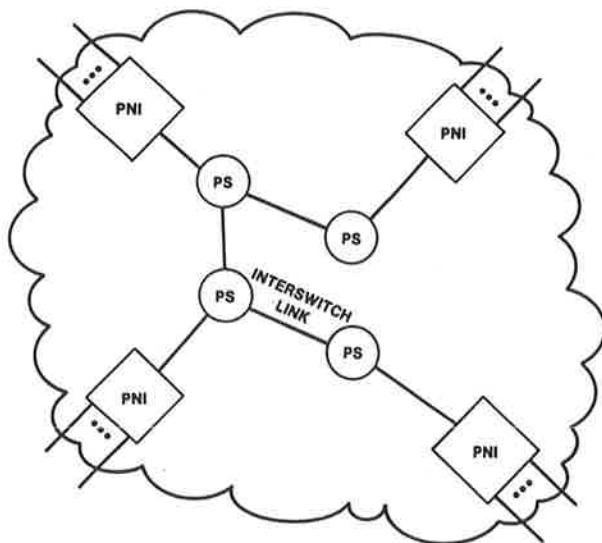
1) Packet network interfaces (PNI) through which users (i.e., subscribers and other networks) connect to the network.

2) Interswitch links that carry voice, data, and signaling information.

3) Packet switches (PS) that connect to both packet network interfaces and other packet switches with high-speed lines.

Manuscript received April 1, 1983; revised July 1, 1983. A portion of the paper was presented at GLOBECOM '83, San Diego, CA, November 28–December 1, 1983.

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PS: PACKET SWITCH  
PNI: PACKET NETWORK INTERFACE

Fig. 1. Packet network components.

### III. FACTORS INFLUENCING THE PROTOCOL DESIGN

Several factors, all interrelated, have a pronounced effect on the protocol design. The primary underlying philosophy behind the design of a voice/data packet network and protocol is to consider these factors and design the network to take advantage of them. Three factors have been identified that significantly affect the packet voice/data protocol: characteristics of the applications using the network, performance requirements, and transmission facilities.

#### A. Characteristics of the Application

The transport of packetized voice is a major application of an integrated packet network. Because of the redundancy in this voice information, it may not require the robust error correction methods commonly found in packet protocols. Exercising conventional flow control on packets containing voice information seems inappropriate. Due to delays that can be introduced through the application of flow control procedures, voice information sent through the network may be rendered unusable. Degrading the level of service for established calls, a characteristic of such flow control procedures, is an undesirable practice in a voice network. Additionally, flow control practices have been developed to allow communication between devices operating at different rates. Because voice coding and decoding equipment would be operating at the same rate, such flow control practices are not necessary. This does not preclude the need for regulating the flow of information into the network to protect network resources.

#### B. Performance Considerations

The design objectives for a packet voice network place several constraints on the operation of the network. Two categories are particularly influential in the protocol design: loss performance (lost data/voice packets, lost signaling packets) and delay objectives [11].

Delay objectives for the network can be met, in part, through the elimination of *link* error recovery techniques that require retransmission of corrupted information. This has a twofold effect: first, one source of variable delays introduced by packet networks is eliminated, and second, network transmission procedures are simplified.

#### C. Quality and Speed of Transmission Facilities

The exclusive use of high-quality, high-speed (e.g., over 1 Mbit/s) digital transmission facilities in a packet voice network [1] encourages examination of nontraditional error correction techniques. Current methods, aimed at ensuring error-free information delivery, typically correct errors on each link. While this complexity and expense (in elapsed time, processor resources required, memory, and buffer utilization) can be justified for slow transmission facilities plagued with "high" bit error rates, the supporting reasons become less clear for the high-speed transmission facilities that would be used in a packet voice network.

The performance of the interswitch digital transmission facilities can be expected to provide sufficiently reliable performance to allow the responsibility of error correction to be moved to the packet network interfaces. As a higher level function, the error correction required for data communication would not be performed on each network link, but once across the network. This simplifies processing that must be done by network switches without degrading the performance seen by network users.

### IV. NETWORK ACCESS ARRANGEMENTS

The placement of functions (e.g., voice encoding) in a packet voice/data network is specified by the protocol architecture. For the generic packet voice/data network considered in this paper, two alternatives are possible for voice coder placement. In the first, the interface to the network is a circuit mode interface. The voice stream into the network is a deterministic stream of information. At the PNI, this stream is encoded, placed in packets, and transported across the network. At the distant edge of the network, the packet stream that was carried through the network is converted back into a circuit voice stream. The data interface to the network is packet mode, possibly using the X.25 protocol.

The alternative method uses a packet mode interface to the network for both voice and data traffic. Voice encoding and packetizing functions are performed by customer equipment, before information enters the network. The network does not process (e.g., code and packetize) the voice information, as it did in the first case, but only transports packets between network customers. As with the first type of interface, two grades of network transport are required: voice transport (fast, but errors are permitted) and data transport (error-free service is required). Other grades of network transport may be desirable, but this paper limits discussion to only these two possibilities.

The remainder of this paper focuses on a circuit-mode

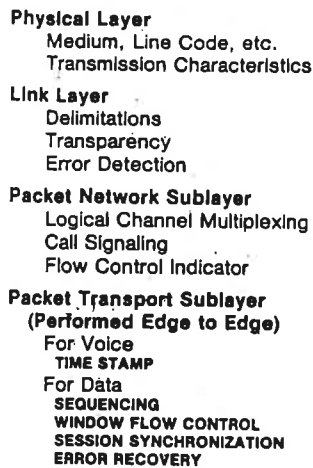


Fig. 2. Protocol functions.

voice interface to the network and a packet-mode data interface. Under this architecture, the packet network interface performs voice encoding and packetizing at the edges of the network. Although this discussion concentrates on internal network operation and functions, these concepts can also be extended, migrating functions from the network to customer premises equipment.

## V. A VOICE/DATA PACKET PROTOCOL

Interswitch communication in a voice/data packet network is according to the network transport protocol. Access to the network is not under the protocol specified for interoffice transmission.

Following the spirit of the draft open systems interconnection (OSI) model that the International Organization for Standardization is formulating [12], this partitioning of functions and rules is specified as a hierarchy of layers, each performing specific services, such as error detection. The packet transport protocol outlined here has been designed using the general principles that have been guiding the development of the OSI model. This voice/data packet protocol supports only transport functions (i.e., the transport of packets through the network); the higher level functions (end-to-end) discussed in the OSI model appear as higher layers above this basic transport capability and are not discussed here.

The protocol structure provides the functions necessary for flow control of data within the network and ensures reliable data transmission through the network. Under this structure, the network can be a reliable data-transfer medium. The basic transport mechanism is provided by the lower three levels of the protocol (1-, 2-, and 3-lower) applied to transmissions across each network link. The packet transport sublayer (level 3-upper), employed edge-to-edge within the network between packet network interfaces, provides functions specific to the application (Figs. 2 and 3). In this scheme, the lowest three levels provide packet transmission and routing through the network, but allow the possibility of packet corruption and packet loss. At the edges of the network, any detected errors are corrected with a peer communication between the entry

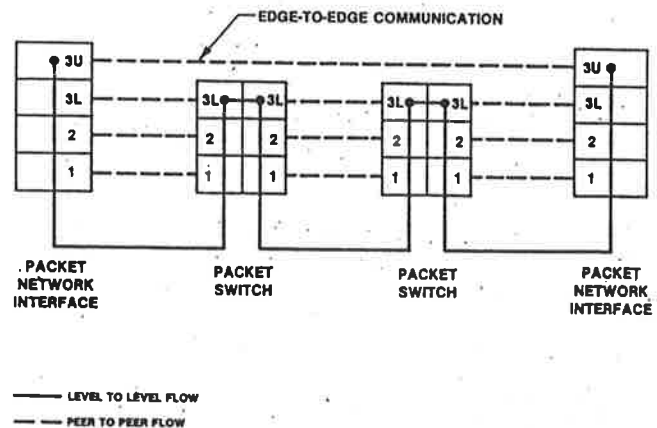


Fig. 3. Internal network protocol layers.

and exit packet network interfaces, thus providing reliable data communication through the network. This protocol structure allows the same link communication protocols to be used for voice and data, and employs an edge-to-edge protocol that is tailored to the specific service used by the network customer. A later section discusses delays experienced by data using this edge-to-edge recovery strategy.

Controlling the flow of information into the network can be accomplished with several mechanisms: call blocking if insufficient network resources are available at the time the call is requested; packet discarding during short periods of congestion; call clearing by the network to face occasional overloads caused by failures of network components (links, switches, etc.); window flow control for data; and other methods.

The following sections present a closer view of the each protocol level.

### A. Level 1 — Physical Layer

The physical level of the protocol specifies the electrical characteristics and representation of transmitted bits. Additionally, bit time synchronization and performance characteristics are included in this level. Level 1 is, in practice, similar to the level 1 of conventional packet data protocols and is not discussed further here.

### B. Level 2 — Link Layer

The link layer performs several functions necessary for successful transmission between network switches. These functions include frame delimiting, error detection, and bit pattern transparency. As a major departure from HDLC, error recovery procedures are not included in this level. Functions such as link set up, link disconnection, and link resynchronization procedures are not included in the packet voice/data level 2 protocol because no link state information is maintained at level 2.

The link layer protocol procedures for the packet voice network do not include provisions for error recovery or for flow control. When an error is detected in a frame, the corrupted frame is immediately discarded by the switch. No further actions are taken at level 2. These measures are aimed at reducing sources of network introduced packet delay.

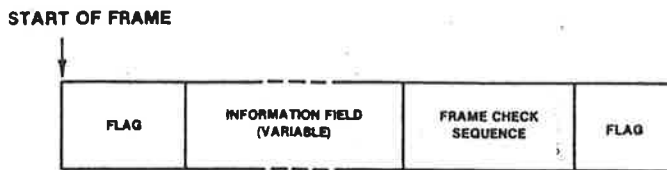


Fig. 4. Level 2 frame format.

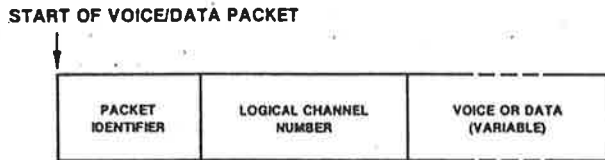


Fig. 5. Level 3-lower packet format.

Each level 2 frame carries a data, voice, or signaling message. The format for information frames (Fig. 4) includes a leading flag that delimits the start of an information frame, a variable-length information field, a frame check sequence for detecting frame errors, and a trailing flag that delimits the end of the information frame. A two byte frame check sequence could probably provide adequate error detection capability.

### C. Level 3L — Packet Network Sublayer

Level three of the voice/data protocol has two sublayers: level 3-lower (3L) and level 3-upper (3U). The packet provides the unit of information transfer at the packet network sublayer of this protocol. The primary function provided here is the routing of packets along a path that is established at call-setup time. This path is fixed for the duration of the call.

Two types of packets are included in the level 3L portion of the network protocol: signaling packets and data/voice packets. Signaling packets are used in call establishment, call disconnection, call administration, and link maintenance. These signaling packets are exchanged between adjacent switches (link signaling packets) and between packet network interfaces in the call path (edge signaling packets). Data/voice packets carry information being sent between level 3U entities. Fig. 5 shows a possible format for a voice/data packet. Information contained in the level 3L header includes three fields: a packet identifier field indicating the type of packet and processing required by the switch, a logical channel number field that uniquely identifies a call on a link, and a variable-length voice/data field containing higher level information. Signaling packets would have similar format, with a signaling message indicator and additional parameters carried in the variable-length voice/data field.

### D. Level 3U — Packet Transport Sublayers

Multiple types of sublevel 3U allow the network to support different classes of packet transport. The specific messages exchanged and actions taken at level 3U depend on the application being supported by the network (e.g., packet voice transmission and interactive data exchange).

For packet voice communication, level 3U functions are performed at the voice coder location (in the packet network interface). Similarly for data exchange, packet transport sublayer functions are also carried out between packet network interfaces (as an edge-to-edge protocol). However, the concepts presented here do not preclude alternative architectures that place these functions external to the network.

There are two distinct sets of level 3U functions: those for reliable data communication and those for voice communication.

1) *Packet Transport Sublayer for Packetized Voice:* For packetized voice only, level 3U messages provide synchronization of voice information. Voice that has been encoded and packetized must be time identified, in some manner, relative to either an absolute time reference or to other voice packets (both preceding and subsequent) to allow a smooth play back of voice at the receiver. This level 3U function provides a means of playing out, at the correct time, received voice information. Such a service is necessary because of the variable delays that can be encountered in a long haul packet network. Voice messages are never retransmitted with this protocol architecture.

A time stamp placed in the level 3U header could identify, relative to other voice messages for the call, the time at which the voice message was generated. The information contained in this field would be generated and used only by the two packet network interfaces in the call path. Tandem switching equipment does not include level 3U processing, so this field would not be modified by intermediate switches. Other time stamping schemes are possible [13].

2) *Packet Transport Sublayer for Reliable Data Communication:* For data communication only, level 3U includes the functions necessary to provide reliable, error-free data transmission. These functions include synchronization of level 3U entities, flow control, and detection and recovery of lost messages. Protocol standards, such as HDLC and CCITT Recommendation X.25, have evolved with the intention of providing reliable communication between devices that do not necessarily process data at the same rate. The packet voice/data protocol supports these functions at level 3U; the packet transport sublayer for data assures error free, sequential message delivery and includes provisions for flow control.

A possible message format for the packet transport sublayer, shown in Fig. 6, includes a message format (MF) field consisting of subfields that can indicate the logical relationship of sequential messages, use of normal or extended-sequence numbering, and other functions necessary to support the access protocol for data; a control field containing flow control and check pointing information; and a variable-length data field.

The set of supervisory messages includes messages necessary to establish and synchronize a level 3U data connection, reliably transfer data across the network, and clear the connection. Level 3U procedures might be thought of as a modified (and enhanced) set of HDLC control procedures exercised across the network (rather than across a single link).

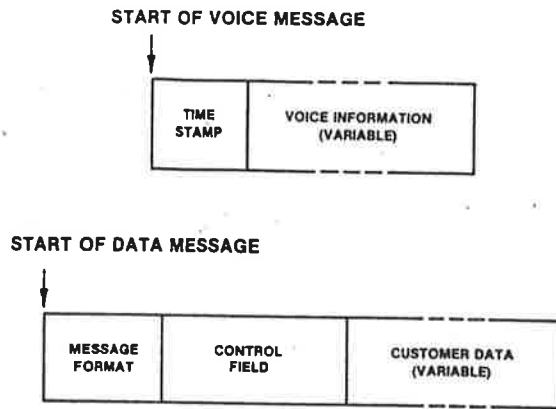


Fig. 6. Level 3-upper message formats.

## VI. DELAY WITH RETRANSMISSIONS FOR CORRUPTED DATA

This section examines cross-network delay using two different error recovery disciplines. The first uses retransmission between adjacent network switches to recover corrupted data; the interswitch protocol guarantees reliable transmission across each network link. Average cross-network data transmission delays for this protocol are compared to the delays encountered under a second protocol. In this second protocol, lost or corrupted data are recovered by edge-to-edge retransmission procedures between the sending and receiving packet network interfaces (this is the method used in the voice/data packet protocol previously described); the interswitch protocol makes no attempt to recover corrupted data.

### A. Link Recovery Model

This section describes a model for analyzing performance using a reliable link protocol to accomplish error-free transport across a network, with a concatenation of reliable links forming the error-free path.

A frame sent across a link may be lost with probability  $p$ , which is determined by the error characteristics of the transmission facilities and the size of the frame sent across the link. If this happens, the link is effectively out of service until the sender "pulls back" and retransmits the corrupted frame. The cycle time,  $T_C$ , is the elapsed time before the sender realizes that it must retransmit the corrupted frame. The cycle time has three components: 1) the time for the frame to cross the link, 2) the interframe arrival time (i.e., the time between reception of the corrupted frame and reception of the next error-free frame), and 3) the time for the reject to be returned to the sender. During this period, the receiver discards all frames it receives. To frames awaiting transmission, the link appears to be transmitting a single, long frame (called an *Xframe* in this paper) during the cycle time. Note that this *Xframe* is not an actual frame transmitted on the link, but serves to describe the amount of time the link is unavailable because of error recovery procedures.

As a preliminary effort to gauge the effect of the link unavailability during the cycle time, the link is modeled as an  $M/G/1$  queue. For this model, link traffic has two components: frames containing useful data (which also

contain embedded data acknowledgments for traffic in the reverse direction), and *Xframes*. Using an exponential service time distribution for user data and a deterministic service time for *Xframes* (this service time is exactly equal to the cycle time) completely describes all traffic on the link for the model.

Using an  $M/G/1$  queue to approximate the average transmission time per link for data ( $T_L$ ), a weighted first and second moment ( $\bar{x}$  and  $\mu_2$ ) describes characteristics of the link traffic.  $D_{prop}$  is the propagation delay per link and  $\rho_L$  is the total utilization of the link; the utilization of the link attributed to data traffic ( $\rho_{data}$ ) is slightly lower than  $\rho_L$ .

$$T_L = (D_{prop}) + \bar{x} + \frac{1}{2} \frac{\rho_L}{2(1-\rho_L)} \frac{\mu_2}{\bar{x}}$$

The average cross-network delay for a path of  $N_{links}$  links, for frames containing data, is

$$\text{Average Network Delay} = (N_{links})(T_L). \quad (1)$$

This delay equation will be used later in comparing edge-to-edge and link recovery strategies.

While the effect of *Xframes* can be negligible at low link speeds, it can become significant at higher link speeds (e.g., over 1 Mbits/s) because of the increasing size of the *Xframe* and the number of frames that must be retransmitted. For example, at 75 percent, total link utilization on a 1.5 Mbits/s link, each frame error causes the link to be unavailable for useful data transmission for over 30 frame transmission times, creating a delay for subsequent frames. The probability of the link corrupting a frame closely describes the amount of link traffic attributed *Xframes*. For the model, every time a frame is corrupted this *Xframe* appears on the link. Because a frame may be retransmitted several times (additional retransmission is necessary if the frame becomes corrupted when retransmitted), an attempt to transmit a useful data carrying frame across a link causes  $p/(1-p)$  *Xframes* to appear on a link. This information is used in determining the probabilities of normal frames and *Xframes* being present on a link, which in turn is used to find the weighted  $\bar{x}$  and  $\mu_2$ .

### B. Edge Recovery Model

This section describes a model for analyzing performance in a network that uses a nonguaranteed-delivery link protocol, supplemented with a reliable edge-to-edge protocol across the network for error-free packet transport. With an edge-to-edge strategy, error recovery is between the corresponding edges of a call, and is done on a per call basis. In this section, frames are sent between adjacent switches on a link. Corrupted frames are simply discarded when a switch detects an error. Messages are sent between corresponding edges of a call. If a message is lost in traversing the network, it is retransmitted again from the sending edge.

The probability of having to invoke error recovery procedures is based entirely on the probability of successfully crossing the network. We find this using the probability  $p$  of a frame being corrupted on a link. From this the

probability  $P$  of, anywhere in the network path, corrupting a frame containing this message is

$$P = 1 - (1 - p)^{N_{\text{links}}} \quad (2)$$

Each link carries a mix of three types of traffic (as contrasted to the link model, in which each link carried only two types of traffic: information frames, which carry customer data, and Xframes). The first type is data traffic, which has the same exponential distribution that is used for the link scenario. To allow a fair comparison with the link recovery model, the amount of traffic attributed to user data,  $\rho_{\text{data}}$ , is identical for the two models. For the edge recovery model, every data message has an explicit acknowledgment, which has a deterministic service time distribution. This is the second type of traffic. Discarded message traffic enters in as the third factor contributing to link utilization. Discarded messages that the sender must retransmit have exponentially distributed service times. The average service time for all traffic on a link is a weighted average of these three service times. The total link utilization for this model,  $\rho_E$ , reflects contributions from each of these three generators of link traffic.

In a call between transmitting edge  $T$  and receiving edge  $R$ , the delay between the occurrence of an error and discovery of this error by  $T$  has three components. These are analogous to the three outlined for link recovery: intermessage arrival time (the time between the moment the corrupted message would have arrived and the time the next uncorrupted message arrives), successful cross-network transit of a data message, and successful cross-network transit of the reject message in the reverse direction.

For a single call, receiver  $R$  discards  $N_{\text{discarded}} = (T_C)(H) + 1$  messages if the sender continues transmitting messages after the error occurred.  $H$  is the user throughput class expressed in the message per second and  $T_C$  is the edge-to-edge cycle time (as defined in the link model discussion). Because of retransmissions, the event of corrupting a message can happen more than once in trying to get a message through the network. Each message sent through the network results in  $(P/(1-P))N_{\text{discarded}}$  extra messages. These extra messages directly increase the queueing delays experienced packets for other calls in the network.

Recall the definitions for  $\rho_L$  and  $\rho_E$ .  $\rho_L$  is the utilization of each network link in a network using link recovery.  $\rho_E$  is the utilization of these links if the network uses edge-to-edge recovery. In general, for a fixed data traffic load (i.e., fixed  $\rho_D$ ),  $\rho_E$  is greater than  $\rho_L$ . This is for two reasons: the explicit data acknowledgments under the edge-to-edge protocol (that are not present in the link scenario) and the comparative error rates (2).

1) *Cross-Network Delay (Including Effects of Retransmission Traffic)*: Computing the average transmission time across a link for data ( $T_{\text{data}}$ ), including propagation delay and system time,

$$T_{\text{data}} = (D_{\text{prop}}) + \frac{1}{\mu_{\text{data}}} + \frac{1}{2} \frac{\rho_E}{(1 - \rho_E)} \frac{\mu_2}{\bar{x}}$$

In this equation,  $1/\mu_{\text{data}}$  is the service time for frames carrying user data; and the first and second moments ( $\mu_2$

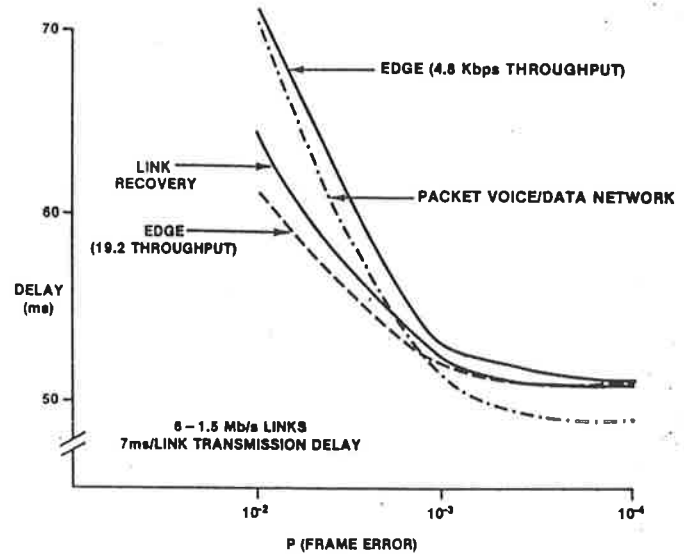


Fig. 7. Average cross-network data delay.

and  $\bar{x}$ ) are found using the service time distributions of the three types of link traffic along with their probabilities of occurrence.

The total average delay incurred in successfully traversing the network one time (without error recovery procedures) for data is  $D_{\text{data}} = (N_{\text{links}})(T_{\text{data}})$ .

The average intermessage arrival time ( $I_M$ ) is based entirely on the user transmission speed and amount of data that can be transported in a frame, causing an increase in error detection time at the receiving edge as user throughput class decreases.

For edge-to-edge error recovery, the equation for cross-network delay, including the effects of retransmission procedures, is

$$\text{average cross network delay} = \frac{(1 + P)D_{\text{data}} + PI_M}{(1 - P)} \quad (3)$$

This delay equation will be used later in comparing edge-to-edge and link recovery strategies.

### C. Delay Comparison

Fig. 7 shows comparative delays for the two recovery disciplines in two types of networks, found using (1) and (3). These delays are for a cross network path of 6 1.5 Mbits/s links, a 7 ms/link propagation and processing delay, uncorrelated frame errors, and frames that contain 128 bytes of customer data not including the frame header. This analysis is not all encompassing; the intent here is to show the effect of Xframes when high-speed facilities are used and show the effect of user throughput on average data delay when using the edge-to-edge recovery mechanism.

In Fig. 7 the amount of data traffic carried is the same for the link and edge recovery curves, ensuring equivalent traffic conditions for the comparison. However, this does not imply that the total link utilization is the same for the two models. (Less data traffic is offered to the voice/data packet network because that network must also carry voice packets.) The method used to generate the link utilizations forces the total link utilization for the link recovery method

to be constant, independent of the probability of corrupting a frame. This means that as the probability of corrupting a frame decreases (moving from left to right in the figure), the ratio of useful data traffic to Xframes increases, and the amount of data traffic increases. For the two edge recovery curves, the amount of useful data traffic increases to stay the same as for the link curve.

Fig. 7 shows that the delays using link recovery are close to the delays encountered using edge-to-edge recovery under the same data load. Not unexpectedly, as the probability of corrupting a frame decreases, the average delay decreases, because fewer retransmissions are required for either method. In the link case, the Xframes caused by the retransmission procedures are large and could add significant delays when they appear on a link. With fewer retransmissions (i.e., fewer Xframes), the total link utilization decreases, diminishing the magnitude of the queueing delays. For the edge-to-edge curves, a major component of cross-network delay is the time to notify the sender that a message must be retransmitted. With this occurring less frequently, the average cross network delay decreases. Additionally, less "extra" traffic (that must be retransmitted) is now appearing on the link, decreasing the queueing delays. However, this is a minor effect for low user throughput classes.

As user throughput increases, two factors affect the amount of "extra" messages present on the link for the edge-to-edge case. First, the number of these extra messages sent during the cycle time (i.e., time to notify the sender that an error has occurred) increases simply because more messages can be sent per second. Along with this, the intermessage arrival time decreases, causing the cycle time to decrease. The net result is that as throughput increases, the number of superfluous messages increases, and for a fixed data load this slightly increases the queueing delays. However, the decrease in time to detect the error significantly decreases the delay introduced by the retransmission protocol. Fig. 7 shows this in the two edge curves, demonstrating that the faster user throughput (19.2 kbits/s) has a lower average delay than the 4.8 kbits/s user throughput. (Note that for user throughputs well over 100 kbits/s, these superfluous messages can cause the link utilization to exceed available bandwidth if the offered data load is held constant. Factors such as link speed and error rate determine the actual customer bandwidth at which this occurs.)

The curve for the voice/data packet network shows average data cross-network delays for an integrated packet network that uses edge-to-edge recovery for data. Although the total link utilizations for this curve are comparable to the other curves, the amount of data traffic is less for this curve because voice traffic is also carried. Corrupted data introduces some extra traffic that must be retransmitted, but corrupted voice messages do not cause any additional network traffic. Because of this, the amount "extra" link traffic caused by retransmission procedures is not as great as it is for the two data-only networks. For this example, frames containing voice messages have a length of 250 bits with a deterministic distribution and outnumber useful

data messages by 4 to 1. This ratio does not include discarded messages or acknowledgment messages.

#### D. Extending the Models

While this paper shows several concerns in finding delay and throughput for the two recovery methods, more can be done in this area. Possible efforts aimed at continuing this study include modifying the model to more accurately describe the probability of corrupting a frame. Additionally, including the effects of corrupted acknowledgment and reject packets will give a more accurate description of network traffic and cross network delay. An enhanced model can include provisions for various message length distributions, increasing the usefulness of the results.

#### E. Conclusions for the Delay Models

For link correction, two factors influence the effectiveness of the network: 1) unless the network uses selective reject, a frame error *prevents* the sending party on a link from transmitting useful frames for a short period, increasing the cross network delay for subsequent frames; 2) the amount of time required to detect and correct a frame transmission error depends on the link speed and utilization.

When using edge-edge recovery, network protocols are characterized as follows. First, when a message is lost, this prevents useful messages from being sent *for a single call* until the sender retransmits the corrupted message. This causes extra link traffic, increasing queueing delays for all calls on that link. Second, the amount of time it takes to correct for a message error is dependent on the user throughput class. (This does not include errors that are detected by the expiration of a timer.)

In a packet network that sends 128 byte messages as a single packet through the network, link recovery and edge-edge recovery methods give comparable performance as seen by a network customer. This conclusion is valid for networks using high speed transmission facilities (e.g., over 1 Mbits/s) having good error characteristics (e.g., corrupted packet rate of  $10^{-3}$ ) with reasonable link utilizations for data traffic (e.g., less than 60 percent).

In a combined voice/data packet network, an edge recovery mechanism can provide acceptable delay characteristics for data traffic.

### VII. SUMMARY

Several concepts outlining the philosophy of a possible protocol structure for integrated voice, data, and signaling have been presented. The protocol overview covers the rules for communication between adjacent network switches, and across the network between the corresponding packet network interfaces of a call. In this protocol, error correction procedures are not included at the link layer or packet network sublayer for data or voice information. Error recovery for data is performed edge-to-edge, across the network. The average cross-network delays for

data packets under this edge-to-edge error recovery discipline are comparable to the expected delays under a more traditional link-by-link recovery protocol.

While this paper has centered discussion on a possible design for a voice/data *internal* network protocol, the area of accessing such a network with an integrated packet protocol is ripe for study. The concepts outlined in this paper of providing basic transport functions at levels 2 and 3L in the protocol, and tailoring the packet transport sublayer (3U) for the specific application (e.g., packet voice) can be extended to apply to network access protocols.

#### ACKNOWLEDGMENT

The material presented in this paper has emerged from discussions with members of the Advanced Switching Technology Laboratory at Bell Telephone Laboratories. M. Decina (while at Bell Laboratories on leave from the University of Rome), J. Turner, W. Montgomery, and L. Wyatt have provided their time in discussing packet voice communication, and their help is appreciated.

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## A Distributed Experimental Communications System

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IEEE Journal on Selected Areas in Communications,  
Vol. SAC-1, No. 6, December 1983,  
Pages 1070 - 1075.

# A Distributed Experimental Communications System

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**Abstract**—The packet experimental communications system (packet XCS) is a new experimental voice and data switch. It uses a local-area network (LAN) for digital voice transmission, with local intelligence for switching. The packet XCS also has highly distributed control. The individual sites cooperate to provide user services as well as internal data management.

We have learned that several local networks, including CSMA/CD networks, can be made to work well for voice transmission and that highly distributed control is practical in such a system. A system has been constructed which is used as a testbed for distributed voice and data communications experiments.

This system is purely for experimentation and does not indicate a direction for future Bell System product offerings.

## I. INTRODUCTION

WE have recently designed and implemented a new voice and data switch for experimental use. Our switch, the packet experimental communications system (packet XCS), uses a local-area network (LAN) for transmission of packetized digital voice among the voice sites. We have thereby eliminated the central switch. The sites have individual controllers to access the LAN and local intelligence to support switching. We have also eliminated centralized control and data management. The packet XCS distributes these functions to the individual sites, which transmit and receive control and data messages over the LAN. In short, the packet XCS uses a shared transmission facility but has distributed switching, distributed control, and distributed data.

While transmission and switching of data have historically been provided on top of voice-oriented mechanisms, recent innovations and cost reductions in digital networks make it possible to implement voice on top of data. The most rapidly evolving field in data communications is LAN technology [2]; high bandwidths can give reasonable voice capacity, recent VLSI support has made LAN's less expensive, and emerging standards promise to aid in planning. The design of the packet XCS is relatively independent of the LAN used (which need only support point-to-point and broadcast packet transmission) although performance is affected by the choice of LAN.

The packet XCS is an experimental system, and this has heavily influenced its design. It has served as a testbed for distributed systems research. Our goal of a highly distributed packet XCS has led us to many interesting theoretical

issues. We have had an opportunity to consider the effect of alternative architectures. For example, we achieve reliability by the packet XCS's lack of central failure points rather than by their duplication. Now that an early implementation exists, the packet XCS seems natural for supporting ongoing research in integrated voice and data services. In the end, the packet XCS may point the way to a useful design in its own right, with essentially zero initial cost and constant incremental cost per site.

## II. TRANSMISSION AND SWITCHING

The packet XCS incorporates shared transmission and distributed switching. Voice packets are transmitted over the network by the voice sites; local switching functions at the sites implement voice paths and other switching abstractions.

### A. Transmission

The selection of a LAN for the packet XCS can greatly affect the performance, cost and reliability of the system. The only constraint we place on the LAN is that it be capable of broadcasting a message to all sites on the network. This is required by several of the switching algorithms presented in the next section. We looked for a number of qualities in a LAN. Since the failure of the LAN would disrupt an entire XCS, the chance of such a failure should be small. The network should be able to carry enough voice traffic to make construction of a medium size packet XCS possible. In addition, a moderate amount of data capacity should be available.

There are several LAN technologies that meet our constraints. Many types of ring networks would be suitable, since they broadcast efficiently [5], [11], [15], [17]. Some of these networks, however, have "head-end" hardware which arbitrates access to the network. This is less desirable than a distributed contention mechanism, since a failure in the head-end could bring down the whole network. Other ring-like networks such as FASNET and the Zurich ring have distributed contention which lessens this problem [5], [12]. Another interesting class of LAN's are the CSMA/CD networks [10], [11], [14]. Since CSMA/CD networks are based on a shared broadcast media, they meet our prime constraint. Our experience indicates that they are very reliable.

For our experimental packet XCS, we chose the CSMA/CD network defined in [9]. This network has

broadcast capability, is reliable, appears to have enough capacity to meet our needs, and has commercially available implementations. This network is also being considered for standardization by the IEEE 802 Committee [11].

Several questions about carrying voice traffic on a CSMA/CD network need to be answered. For example, how many voice circuits can be provided? What is the effect of packet size on capacity and delay? How much data will be lost due to excessive delays? And, finally, are there transmission strategies or slight modifications to the contention mechanism that will improve transmission? The next few paragraphs will address these questions.

As a starting point, we assume that speech is digitized using a  $\mu$ 255 PCM coder at 64,000 bits/s. To better utilize available bandwidth, we assume that speech detection will be implemented in the site, and that each site will transmit an average of 40 percent of the time [3]. If the channel capacity is 10 Mbits/s, complete utilization of the bandwidth would result in a capacity of 195 conversations. This is an ideal, since efficiency is decreased by per-packet overhead and contention overhead.

Transmission on the network is bit-serial, beginning with 64 sync bits, followed by the packet. A packet contains 112 bits of header, a data field of between 368 and 12 000 bits, and a 32 bit CRC field. Assuming 64 kbits/s speech, a packet can hold from 5.75 ms to 187.5 ms of speech. A site may begin to transmit when it has seen the network idle for 9.6  $\mu$ s. If voice sites are uniformly distributed along a maximum-length network, computations based on the network propagation delay budget give a worst-case mean one-way propagation time of 10.06  $\mu$ s. We can expect an arbitrary site to see the network go idle 100.6 bit-times after the preceding site actually ceased to transmit. Taking this overhead into account, the capacity of the network is reduced to between 93 conversations (at the minimum packet size) and 188 conversations (at the maximum packet size).

Transmission strategy can have significant effect on the network capacity. Since speech packets are periodic, if two sites' transmissions collide once, they will collide again on the next packet (ideally). This will continue until one of them ceases talking. To avoid this effect, an adaptive packet size is used. A site that experiences a collision will delay and then attempt retransmission. Instead of retransmitting the same packet, any speech samples acquired during the delay will be added to the packet. Since the delays of the colliding sites will be different, their packet sizes will differ, the next transmission times of the sites will differ, and successive collisions will be avoided. An adaptive packet size strategy can increase the throughput of the network by 25 percent or more by reducing the number of excessively delayed packets. The throughputs given in this paper assume the use of an adaptive packet size strategy.

Contention overhead is more difficult to determine. When two or more sites attempt to transmit nearly simultaneously, a collision will result. The rate of collisions and their duration is affected by such factors as the number of sites and their spacing on the network. Since these effects are difficult to describe analytically, a simulation was performed [9]. The result is that 50 conversations can be

transmitted if a packet contains 5.75 ms of speech, and 150 conversations can be transmitted if packets contain 50 ms of speech (larger packets introduce unacceptable delay into the system). At these loads, a very small ( $\ll 1$  percent) number of packets will be delayed so long that they will not arrive at their destination in time to avoid a gap in the conversation. These results are consistent with earlier work [16] but with greater detail.

A CSMA/CD network carrying periodic traffic has a steady-state traffic pattern much like a TDM network. There are few collisions, since each site finds a slot in which it does not collide with other sites. When a new site is added to the network (such as when a talkspurt begins), it may experience a few collisions, which may in turn cause other sites to experience collisions. However, the cascade effect is heavily damped, and steady state is achieved rapidly. This arrangement falls apart in the presence of aperiodic traffic such as data. When a 5 percent data load is added to the simulation, the voice capacity of the network drops by as much as 25 percent. Extra collisions between the periodic and aperiodic traffic cause the drop in voice capacity. However, 5 percent average data is a large load, corresponding to hundreds of medium speed sites, or a few large computers.

Even though simulations show that a CSMA/CD network can carry voice, the stochastic nature of the contention mechanism will cause a very small number of packets to be excessively delayed, even under low loads. Thus, no guarantee can be made that a given packet will make a timely arrival at its destination. In practice, the authors do not believe this will cause significant degradation of a voice channel, unless the network is overloaded. Overloading can be avoided by engineering the packet XCS for slightly less than capacity traffic.

CSMA/CD networks give all sites equal access to the network, and do not favor any given sites. Unfortunately, this is not the optimal strategy in a mixed data-voice environment. In the presence of data, voice should have priority, so that it can meet its real-time constraints. Most data can be delayed with no serious effects. A particularly interesting variation of CSMA/CD, called movable-slot TDM (MSTDM), offers a solution to these problems [13].

The idea behind MSTDM is to allow voice packets to preempt data packets on the network. This is done by having voice packets ignore collisions with data packets. When a voice and data packet collide, the data packet transmission is stopped, and retried later. The voice packet continues, and acquires the channel. Voice packets have a "preempt header" that can be garbled by collisions with no ill effects. Collisions between voice packets are impossible, due to the nature of the access mechanism. The important thing about MSTDM is that it places an upper bound on the delay of a voice packet transmitted through the network. Thus, no voice packets will be lost due to excessive delays.

In his paper, Maxemchuk [13] considers a CSMA/CD network that has different parameters from the one we have been considering. It operates at 3 Mbits/s, and has much lower per-packet overhead. In addition, the voice is encoded at 32 000 bits/s. These parameters are picked to

increase the efficiency of the network. If one applies the MSTDM technique to the 10 Mbit/s CSMA/CD network we have been considering in this paper, a considerable reduction in efficiency might be expected. However, preliminary calculations show that MSTDM operating on our network permits a voice capacity almost identical to pure CSMA/CD. In pure CSMA/CD, channel capacity is lost to collisions. In MSTDM, no voice collisions are possible, but voice packets have an additional preempt header one slot time (512 bits) long, which approximately negates the advantage gained by collision reduction.

In summary, it is possible to transmit reasonable numbers of voice channels on a standard CSMA/CD network, with very low losses. If the MSTDM technique is applied, no voice packets will be lost due to the contention mechanism.

### B. Switching

Switching is performed at each site. There is no crucial centralized switching hardware. Peripheral control software implements a voice protocol layered above the underlying LAN protocol. Peripheral control also includes local primitives for providing station or trunk tones and receiving station or trunk inputs (switchhook and keypad, supervision and tones). These operations are supported by local hardware.

Real-time constraints keep us from using a complex protocol in the voice paths. By the time the loss of a voice packet could be noticed, it would already be too late to retransmit it, and modifying timing constraints to allow retransmission would do more harm than good. Our voice protocol is unidirectional and simply time stamps voice packets; we depend on known network transmission characteristics to achieve good results.

Switching voice paths involves buffering and transmitting speech samples from the mouthpiece and receiving, uniformly delaying, and playing back speech samples into the earpiece. Peripheral control implements abstract operations on voice connections. For a two-site connection, peripheral control would transmit voice packets point-to-point. For a multisite connection, peripheral control might transmit the voice packets point-to-point or to a multicast address. Each site involved in a multisite connection receives the other's packets and reconstructs them into a single stream for playback; although the processing power required is potentially unbounded as the size of a connection grows large, intuition tells us that the instantaneous number of speakers will probably be small and that extreme cases can be allowed to result in some lost speech.

## III. CONTROL

Given the underlying approach to transmission and switching in the packet XCS, there were many possible approaches to the design of the system's higher-level feature control functions, with each choice having a major

effect on the resulting system. Accordingly, we first chose a set of guiding principles which then directed our further design. Our decisions were often arbitrary but were ultimately interdependent; together, they helped to narrow our options and contributed significantly to the conceptual cohesiveness of the packet XCS.

Many control features of the packet XCS could have been centralized or distributed. In each case, we chose the distributed approach as an opportunity to study distributed processing. The distributed approach was also considered a challenge, since voice switching features have evolved in a centralized control environment and it was not obvious whether they could be implemented well, or at all, in a distributed environment. We also chose to distribute all data functions in the packet XCS.

We assumed that the underlying network would be extremely reliable. This reliability, useful in transmission, is also useful at higher levels. Instead of building protocol layers to eliminate lost messages, we decided to live with them. Since a switching system can tolerate some number of failed call attempts, we chose to design our system so that lost messages could result in call failures or degraded service, but in which only the associated call would be affected. This decision requires that messages be associated with only a single call, since losing a global message could have unbounded effect. We also decided to take advantage of the broadcast nature of the LAN for certain data operations.

Sites may fail or come up at arbitrary times. Again, we chose that such downtime and such transitions should affect only calls to and from those sites; this again eliminates having sites with global responsibilities.

We decided that the programming for each site would be specialized. A site supporting a station would contain the code only for that type of station; a site supporting a trunk would contain the code only for that type of trunk. Another consequence of this principle is that trunk selection on an outgoing call should be performed at the trunk sites themselves instead of at the station sites, since the knowledge of the selection criteria should be specific to the trunks.

We decided to rely heavily on automated techniques in the design and implementation of the system. For example, we want lost messages to cause only transient errors; a given design of feature control will have some messages where this is naturally the case but others that must be "reinforced" by extending the protocol. We found that it was possible to distinguish such cases mechanically and to automate the reinforcement.

Some of our philosophical guidelines had to be stretched when special problems arose (in areas as diverse as billing and unassigned numbers), but overall they were extremely useful in helping us focus our design effort.

Feature control for the packet XCS adopts the popular architecture of one process per site. This process executes on a processor at the site, possibly shared with peripheral control. The feature control process contains an extended finite-state control component and a data-management component.

### A. Extended Finite-State Control

The extended finite-state control is a finite-state machine whose inputs are site inputs and control messages from other sites, and whose outputs are site outputs and control messages to other sites. Site inputs and outputs go through peripheral control while control messages are transmitted over the LAN. Each state of the machine accepts some set of possible inputs, each resulting in some list of outputs and a transfer to a new state. For example, from the initial state, the input "off-hook" produces the output "start dial tone" and transfers to a state where the input "digit" produces the outputs "stop dial tone" and "remember digit" and so forth. The extension to the finite-state control is the ability to manage small amounts of data local to the extended machine, such as the digits dialed.

The extended finite-state control is generated mechanically using program synthesis [7]. A specification language allows a nonprocedural listing of facts concerning voice sites (e.g., stations cannot give dial tone when they are on-hook) as well as the features to be provided (e.g., stations should give dial tone when they go off-hook). Our specification language is very close to extended propositional temporal logic [18]. While ultimately equal in power to finite-state machine specification, our specification language seems much easier to use. Independent facts or features can be specified with fewer interdependencies arising as artifacts of the specification. Specifications are mechanically translated to finite-state machine form, with the translator detecting inconsistencies and underspecifications, eliminating some automatically while interacting with the user to resolve the others.

Each site's protocol for interaction with other sites is explicit in the specification. Messages are treated as additional inputs and outputs. Progress has been made toward protocol synthesis, where the specification might describe only the cumulative behavior of a set of interacting sites and the necessary protocol could be derived mechanically. This work is still ongoing; our initial protocols have been constructed manually, with semiautomated checking for protocol bugs. (As with programs, not all protocol bugs can be found automatically, although protocol synthesis can guarantee their absence [8].)

### B. Data Management

Just as switching and control are distributed in the packet XCS, so is data management. For example, translation data (giving the mapping from dialed numbers to hardware site addresses) are not stored centrally; they are distributed among the sites involved. To use the translation data, a site broadcasts a query to all of the sites and the site(s) with the appropriate dialed number responds with its network address. This approach requires only  $O(1)$  (i.e., constant) space per site as a function of the number of sites but requires  $O(n)$  (linear) time per site, with some small constant factor, as the number of sites and, thus, of queries grows.

In general, each site holds those data that pertain di-

rectly to it (thus there are no purely "system" data); the data form data relations and each relation can span many or all sites. When the contents of a relation are desired, the requesting site can broadcast a request for the contents. Each site replies with its tuple(s), and the requesting site performs some operation on these tuples. If a tuple is to be modified, it is first located by broadcasting and then modified using a point-to-point message.

Often, the operation following the query can also be performed in a distributed manner. In the translation data example, the operation is selection of the tuple(s) whose key is the dialed number. The individual sites can perform this selection, with only the site(s) matching the dialed number replying; this reduces the number of replies from  $O(n)$  to  $O(1)$ .

Many such optimizations are possible. A typical operation is to maximize or minimize the value of a key. This is useful in least-cost routing, where we must find the trunk that minimizes some cost; another use is the maintenance of a queue of sites, where finding the head of the queue becomes finding the site with the greatest time in the queue. Here, the first operation is selection of appropriate trunk sites or of sites in the queue; the sites can perform this selection themselves. Next, since the keys are stored only implicitly, they must be recomputed on each access; computing them at the target sites instead of the requesting site is another transformation. Another is having the target sites desiring to reply first listen for other replies and drop out if they see one better than theirs. Yet another is to have the sites's initial periods of listening inversely related to their perceived goodness of their results. Together, these transformations can produce a family of efficient protocols for accessing distributed data [6].

This data distribution technique is only approximate. It can fail if a queried site is down or misses the query or its reply is lost. In the packet XCS this can affect at most the associated call. At best, the resulting behavior can be desirable; if a site in a queue goes down, it will no longer be considered to be in the queue. We rely on the underlying LAN and the relative infrequency of queries to keep the number of lost queries or replies small.

## IV. PROTOTYPE DESIGN AND IMPLEMENTATION

In order to further understand the distributed packet XCS concept, a prototype packet XCS was implemented. It currently consists of a small number of phone sets. A trunk interface is being designed. Plain old telephone service (POTS) for the phones has been demonstrated. This section will discuss our experiences and the insights we have gained from this implementation.

Our first priority was to implement POTS. Our prototype hardware was selected to accomplish this in a timely fashion. Our current phone is based on a PDP 11/23 processor, and consists of six double-height Q-bus boards: CPU, memory, LAN (3), and a voice interface. The LAN interface is the 3Com QE controller [1]. The voice board was custom-designed for this application.

Even though this prototype system uses general-purpose parts, a final implementation of this system should have minimum hardware cost. To meet this constraint, the hardware uses a shared-memory approach. The LAN interface appears to the CPU as an area of main memory into which packets are received and transmitted. The voice interface consists of a codec which loads a FIFO buffer with digitized samples from the mouthpiece, and empties another FIFO filled with samples destined for the earphone. The software is responsible for moving data between the voice FIFO's and the LAN packet memory.

Both peripheral and feature control run on top of a real-time operating system which allows the rest of the software to be structured as a set of cooperating processes. A message-passing IPC mechanism is also provided. The interface between the peripheral control and the feature control is defined by a set of 19 messages divided into four groups: tone generation, user input, switching, and transmission. Tone generation is used to start and stop dial tone, busy signal, audible ringing, and ringing. The peripheral control software generates these tones from tables. User input messages are sent from peripheral control to the feature control when the user dials a digit or takes the handset on-hook or off-hook. Switching messages are messages between the feature control layers of the phones involved in a conversation. They are transmitted, but not interpreted by the peripheral control. For POTS there are only two transmission messages: talk and stop talk, which cause the peripheral control to set up and tear down a simplex path between the local mouthpiece and the remote earphone.

A conversation consists of two independent, identical, simplex voice paths. The algorithms to implement a basic voice path are simple. The transmitter collects samples until a packet is filled, and transmits it over the LAN. The receiver catches packets addressed to it, and plays them out to the phone. This simple algorithm must then be augmented to handle variable packet spacing and speech detection.

There are two unavoidable causes of variable packet spacing: transmission delay and clock drift. Transmission delay is to be expected in a CSMA/CD network. The simulations discussed previously predict that in a pure CSMA/CD network, with 5.75 ms voice packets, 5 percent data loading, and 40 conversations in progress, the mean packet delay will be 0.2 ms, and the standard deviation will be 1.0 ms. If one provides 5.75 ms of artificial delay at the receiver, approximately 1 percent of the voice samples will be lost. With no data loading, the mean and standard deviation of the delay drop to less than one sample (125  $\mu$ s). If MSTDM is added to the network, the maximum delay for a voice packet is less than 350  $\mu$ s under any data load. With at least 350  $\mu$ s of buffering at the receiver, no voice will be lost.

Clock drift is a less obvious cause of lost voice samples. Since the packet XCS is completely distributed, the two codecs involved in a conversation are sampling at a different rate. Over a period of time, the receiver will see either

too few or too many samples. This will eventually cause a break in the conversation. In our implementation, the tolerances of the crystal clocks is  $\pm 0.1$  percent. This means that the receiver can overflow or underflow by as much as 1.6 samples. We currently use 32 ms voice packets, with a total of 64 ms of buffering available in the receiver. If the extra 32 ms of buffering is divided in half to give equal protection against underflow and overflow, a voice sample can be lost in a minimum of 80 s. Our experience indicates that this is a more common cause of lost voice packets than the variable transmission delay caused by CSMA/CD. Only when voice and data loading on the network approach saturation does the transmission delay become predominant. Fortunately, the small amounts of speech lost due to clock drift are easily compensated for. When speech detection is implemented, the receiver can readjust its buffer between talkspurts. Since the average talkspurt is very much less than 80 s, voice samples will rarely be lost.

In order to increase the network capacity to reasonable levels, speech detection is necessary. As each packet is copied from the transmitter's voice FIFO to the LAN buffers, the maximum amplitude sample is noted. If the maximum amplitude sample in a packet does not exceed a threshold, the packet is deemed "quiet." When the transmitter notices a number of quiet packets in a row, it stops transmitting. When it again notices a nonquiet packet, transmission is resumed. The amplitude threshold and number of quiet packets to transmit are fixed. This speech detection algorithm works nicely in an office environment with one exception. When a speaker stops talking, there is a noticeable delay before the transmitter stops transmitting. During this time, all of the background noises in the speaker's environment can be heard. These then abruptly cease when the transmitter stops. The traditional solution to this problem is to have the receiver play white noise into the earphone when the transmitter stops. We chose to implement an alternate strategy, however. The human ear is very sensitive to abrupt changes in amplitude but much less sensitive to subtle changes. So, our transmitter applies negative gain to quiet packets. The  $n$ th packet of a sequence of quiet packets has a gain of  $-6n$  dB. The gain is implemented by table lookup in the transmitter. The result of this is to make the speech detection effect unnoticeable in an office environment. In noisier environments, or with trunks, an adaptive threshold would probably have to be used.

## V. CONCLUSIONS

The packet XCS represents an innovative new design of an experimental switching system. It uses a LAN as a shared transmission facility; it distributes switching, control, and data. A prototype implementation has been constructed and is currently in use.

We have learned that, with proper care, a CSMA/CD LAN can be used as a voice transmission facility. With MSTDM, voice transmission is deterministic while data

transmission retains the desirable CSMA/CD performance. We have also learned that distributed control is possible for such a system, and that various aspects of system design can be profitably automated.

#### ACKNOWLEDGMENT

The authors wish to thank D. Weller and C. Connolly, who were responsible for the design and implementation of the voice board used in the prototype system; J. Lacy, who was responsible for the nonautomatic part of the programming for the prototype system as well as general assistance; N. Maxemchuk for his MSTDM protocol; P. Wolper for discussions on temporal logics; and D. Bergland for his continuing support.

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