Packetizing Voice for Mobile Radio

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Present cellular systems use conventional analog **fm** techniques to transmit speech.' A major **source** of impairment in cellular systems is the Rayleigh fades **[11.[21** that cause the speech signal to be interrupted with noise bursts in the form ***of*** "pops" and "clicks". In digital systems, these fades appear as clustered bit errors. **Since** adaptive digital codes can be designed to be more robust to bit errors, digital *coding* of speech was suggested **as** a possibility. For example, Duttweiler and Messerschmitt [31 **studied** a combination of nearly instantaneous companding ***of*** speech signals and time diversity with parity checks. Variable step-size differential coders based on the forward transmission ***of*** the quantizer step-size information in an error-protected block are described in references [41. Subsequently, adaptive delta modulators were also proposed [5] for mobile telephony.

In all ***of*** this earlier research, **high** bit rate coding ***of*** speech at 24 kbps or higher was used. Subsequently, however, the focus shifted to **high** quality coding at low bit rates. For example, current proposals for digital cellular systems in **the United** States are based ***on*** an **rf** channel bandwdith of 10 IrHz, low bit rate coding of speech on the order of 8 kbps using possibly a hybrid coding algorithm? and a total transmission rate of about 14 kbps [61,M1.3 This results in a better reuse of **rf** channels and an increase in the number of cellular users by a factor of at least 3. For digital cellular systems in Europe, the Group Speciale Mobile is also recommending the use of a hybrid coder operating at a bit rate of 13.2 kbps and a total transmission rate of 22.8 kbps [8].

In this paper, we propose speech packetization at mobiles and the serving tenestrial base stations in cellular systems. Packets are encoded in an error-detecting code. If errors are detected, the packet is discarded and a rebansmission request is generated at the receiving end. *Our* motivation for this approach is the possibility that in this way the receiver may be able to recover a good percentage of the packets that are embedded in fades with a consequent improvement in the quality of the received speech. This is a significant departure from normal procedures because, generally, for on-line packet voice communication on local or long-haul networks where the probability of an error is much less than on a Rayleigh fading channel, packets received in error are neither discarded nor corrected, but me simply played back as received. Even if a packet is lost, there may not be, depending on the packet loss probability, any noticeable degradation in the quality of speech [9]! In mobile telephony, however, things are different. When the rf signal goes into a fade, there is a high probability that digital data bits will be in *error.* And since fade durations longer than 8 ms are not unusual (see section II), speech packets may lbe subjected to long error clusters.

With the approach we suggest, there are some immediate fallouts. First, during an average telephone conversation, speech signals are present only 40% of the time [101,[111. Thus, with speech packetization, one can use a smaller rf bandwidth and further increase the number of cellular users by a factor 2.5 beyond what is achieved with the low bit rate coding of speech. Alternatively, for the same bandwidth, one can send the user data, if any, during silence periods. Furthennore, **since** customer data **is** inherently bursty in nature, it **is** particularly suitable for packetization. Thus, speesh packetization lends itself to straightforward multiplexing with the user data with a single communications pr0toc01.~

Speech packetization is nothing new [121 - [151. In the mid 19$0's, AT&T developed wide-band packet technology to provide simultaneous voice and data in a packet format C161. The system, initially designed for long haul networks, would take ***64*** kbps PCM voice inputs from five Tl lines at the DS1 rate, remove the silence periods, convert the speech samples into 32 kbps **ADPCM,** packetize them according to Certain **rules,** statistically multiplex them and then send them over a single TI line, thus providing a 5:1 bandwidth saving. Reference [17] discusses the issues involved in long haul packet voice communication such as the relative importance of silence detection and speech compression, lost packets, variations of delays in long haul networks, and packet sizes in comparison to high speed local area networks. Integration of voice with data in computer comunications networks and interactive on-line packet voice communication over local area networks have been the subject of research for a long time .

In **this** paper, we present a procedure for transmitting packetized **speech and** show that with a reasonable packet size, it **is** possible to achieve an improvement in the ***SNR*** at the receiv**ing** end by using a simple protocol to provide limited recovery of faded packets. The SNR improvement at the receiver result***ing*** from this procedure is in addition to any improvement that could be achieved with a specially designed coder? Section 2 of **this** paper illustrates the characteristics of a mobile radio channel. Section **3** summarizes some delay-related issues that are important in designing a protocol for a packet voice communication in a mobile radio system. Section **4** describes the proposed packt protocol along with a reassembly procedure that smoothes the variable packet delays in the network and allows voice samples to appear contiguously at the receive decoder. Results are presented in Section ***5.***

CkARACTERISTICS OF A MOBILE W I O ChANNEL

The multipath fading characteristics of a mobile radio channel are well understood. However, for the sake of complete**ness,** we will summarize them here **as** they relate to this paper.

In a mobile radio channel, multipath fading causes the rf signal envelope to vary randomly with a Rayleigh distribution. **A** fade **OCCUTS** when the instantaneous signal falls below its mean. If this happens, noise captures the receiver and the received signal is interrupted with noise bursts. There is another source of impairment in mobile telephony. **In** a mature cellular system, the **rf** channel that **is** used in one cell **is** reused in another that **is** some distance apart. But since **this** distance is finite, during the short-duration Rayleigh fades7 of the signal envelope caused by the vehicle motion, the interfering signal may capture the receiver. The result is a burst of interfering voice which is unintelligible because of **the shd** duration of the fade, but may be frequent and long enough **to** perceptually degrade the quality of speech.

The severity of these noise or interferena bursts depends on the fading rate and the average fade duration which in turn depend, among other things, on the average signal-to- noise (or interference) ratio (i.e. the fade level), **the** vehicle **speed** and the wavelength **of** the carrier. **As** the vehicle **speed** decreases, the number of fades per second for a given fade level decreases; however, in this case, the average fade duration increases. Thus, at smaller vehicle speeds, even though the fades and hence the error bursts OCCUF less frequently, they last longer. **As** a matter of fact, error bursts several milliseconds long are not unusual. For instance, at **850** *MHz* and at 12 mph, the signal goes into a -15 **dB** fade at **the** rate of approximately 6 times a second. The probability that the duration of this fade is **8** ms or more is about **0.2.** Thus if we were to transmit $-bit speech samples every 125 microseconds, a block of ***64*** or more samples would be corrupted with error bursts once every 160 ms with a probability **of** 0.2 when **the** vehicle speed **is** 12 mph and the noise level **is** 15 **dB** below the mean value of the signal. References [21,[4] and **[241** discuss these characteristics in detail. For our study, however, the fade statistics of Table **I** for a fade level ***of*** -15 **dB** are sufficient.

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