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Ian H. Merritt

Providing Telephone Line Access
to a Packet Voice Network

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20. ABSTRACT

Research in the area of packet voice over the past ten years has proven the value of packet technology for voice transmission. Continuing research in this area has led to the development of a packet satellite network for use as a testbed for packet voice telephone services. In order to allow a wide range of users to participate in the testing of this technology, ISI has developed the Switched Telephone Network Interface card to allow interconnection between the packet network and the commercial telephone system. This report describes the design of this device and discusses the techniques used to implement functions such as sound/silence detection, echo suppression, and dialing.

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PACKET VOICE

Digital packet voice has been the topic of a great deal of research and development work at the Information Sciences Institute (ISI) and other research institutions in the DARPA (Defense Advanced Research Projects Agency) community since the mid 1970s. DARPA has provided funding for the development of this technology, which offers the advantages of dynamic routing, excellent transmission quality, and a mixture of voice and data on the same network. It also provides an inherent mechanism for efficient channel utilization by the use of sound/silence detection.

Packet speech was first transmitted over the terrestrial ARPANET. This work proved that the concept of transmitting digital voice signals via packet networks was valid. Special packet network voice protocols have evolved as a result of this work, providing a more efficient transport mechanism for voice signals in a packet environment.

For this new technology to be fully tested, it must be made accessible to a large group of people and put into use on a regular basis. For this purpose, DARPA and the Defense Communications Agency are sponsoring the development of a 3 Mb/s wideband packet satellite network to serve as a voice and data link between sites in several locations around the United States, allowing packet voice technology to be demonstrated and evaluated on a scale closer to that of a real-world application.

To expand the usefulness and flexibility of the new wideband network as a packet voice communication system, it is desirable to provide connections to the commercial switched telephone network (STN) at several geographically separated sites. The STNI (Switched Telephone Network Interface) card [5] pictured in Figure 1 was developed at ISI to provide this connection.

This paper describes the implementation of the STNI card and discusses various aspects of its development, including interface requirements, sound/silence discrimination, user tone signalling, telephone system tone signalling, and disconnect detection.

STNI CARD

The STNI card is designed to provide an interface between the commercial telephone network and a Packet Voice Terminal [7], developed by the M.I.T. Lincoln Laboratory. Figure 2 shows a typical scenario of a call in which the STNI is involved. The voice terminal handles the packet network protocol, providing a digital connection to the network. The STNI card is used to answer calls from the telephone system, present a dial tone, and accept digits from the caller directing a call to be placed in the digital packet voice network. Calls to the switched telephone network may also originate in the packet network; the user routes them to an STNI card and requests it to dial the distant telephone number. Once a call is in progress, the STNI performs the analog/digital and digital/analog conversions between the telephone line signals and network data. The card also performs sound/silence detection as a means of bandwidth optimization.
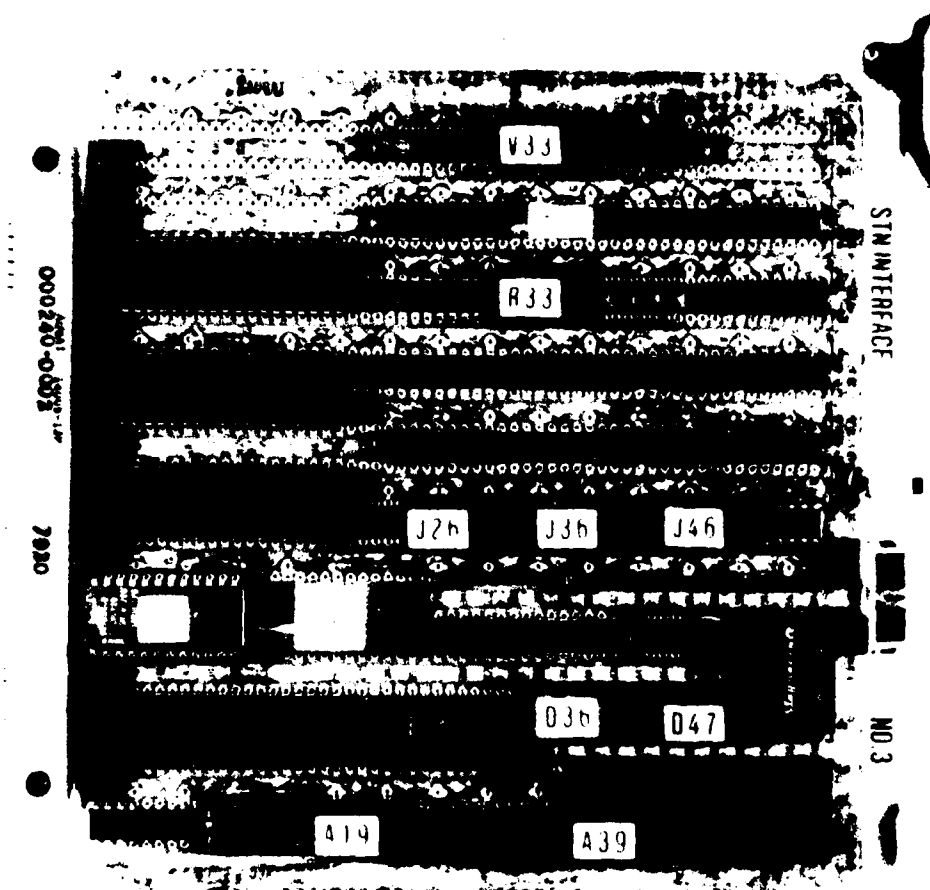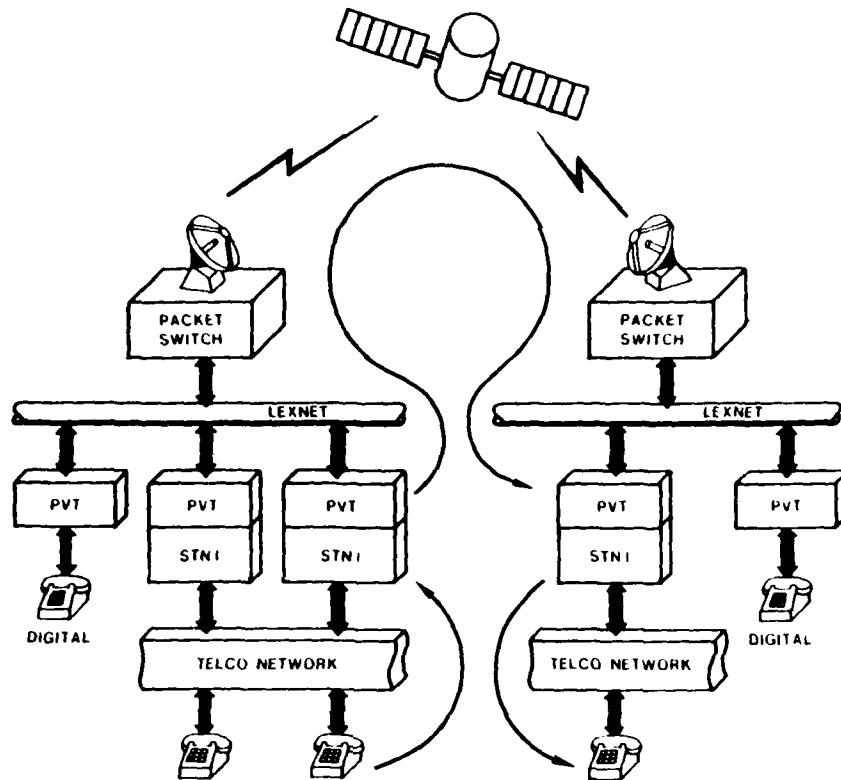
Figure 1: STNI card

The STNI card consists of a Z80 microprocessor with two serial and two parallel I/O channels, an Intel 2910A $\mu = 255$ pulse code modulation (PCM) CODEC paired with the Intel 2912 PCM line filter, a DTMF (Dual Tone Multiple Frequency-touch-tone) decoder, a telephone line ring detect circuit, and a line control relay (Figure 3). Other functions which might have been implemented in hardware are instead performed by software to keep the circuit compact. These include sound detection, echo suppression, disconnect detection, DTMF dialing, and user tone feedback (dial tone, busy signal, etc.). ASCII commands sent by the voice terminal via a 2400 b/s RS-232 serial line instruct the STNI to pick up the telephone line, dial out, hang up, answer the ringing line, and play tones. The same two-way serial line is used in the other direction by the STNI to transmit ASCII status messages back to the voice terminal, indicating a ring-detected condition or received touch-tone digits. This ASCII command port is also usable with a standard serial ASCII terminal and provides an excellent debugging aid.

The arrows show the path of a typical call originating at any telephone. dialing into an STNI card. transmitting up across the wideband network. down to an STNI at another network site. and on to the distant telephone.

Figure 2: Typical use of STNI card

SOUND DETECTION

Perhaps the most critical function of the STNI card. as it interfaces to analog signals, is sound detection. Bandwidth compression is accomplished by taking advantage of silent periods during speech. To provide maximum packet network efficiency without interfering with the clarity of the transmission. the sound detector must be accurate and fast.

Sound detection in the STNI is now accomplished as a software function of the Z80 microprocessor, which analyzes speech data in parcels. each 180 bytes long, representing 22.5 msec. of speech. Statistics are maintained on the data and used by the Z80 program to determine. on a parcel by parcel basis, whether or not a given parcel is silent. Several techniques and refinements, described below, were implemented and evaluated before a suitable method was finally developed.
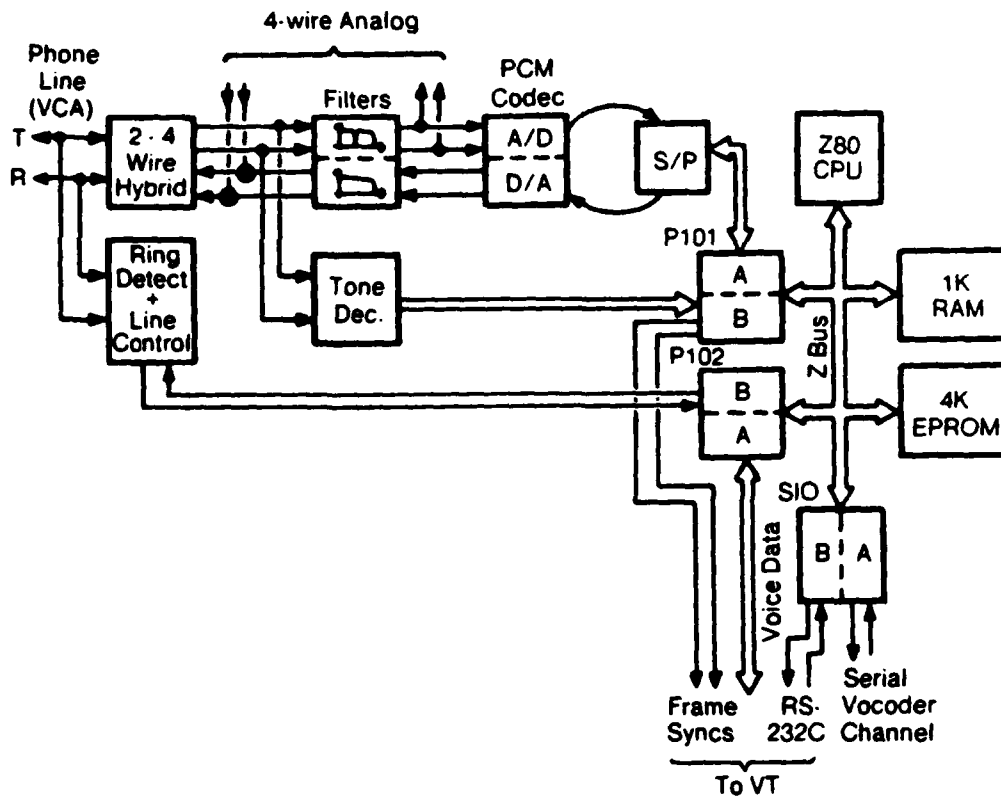
Figure 3: STNI functional diagram

CVSD-Based Sound Detector

Initial plans were to perform sound detection using a continuously variable slope delta modulation (CVSD) encoder on the STNI card. Analysis of CVSD signal patterns should provide a reasonable basis for differentiating sound from silence. This technique has been applied in the past with good results [2].

It was later found that the sound detection function could be performed using the PCM data, thus eliminating the need for CVSD hardware and saving space on the STNI card. As a result, the CVSD-based detector was never fully implemented on the STNI; instead, the hardware was removed from the prototype design.

VOX

In one common sound detector, usually referred to as a VOX or Voice Operated Switch, a fixed threshold and a set of delays provide wide-range sound detection. Sound with amplitude (a) in excess of a fixed threshold (t) for a period in excess of a

fixed delay (d) trips the detector, which is then sustained for a preset hangover time (s). after which it resets. If a exceeds t during the period of s. s is reset to maximum and thus the detector remains tripped.

Among the problems with such a sound detector are the following.

- It is slow to respond. often resulting in the loss of several syllables at the beginning of a burst of speech.
- It sustains longer than is desirable.
- It does not adapt to ambient noise conditions.

A VOX circuit can be adjusted to respond faster at the risk of triggering on noise pulses and clipping soft syllables. VOXs typically do not react very fast. thus consuming excessive bandwidth. They also do not adapt to varying background noise levels. This behavior results in an unnecessarily heavy load on a packet network.

VOX-type detectors are suitable for speaker-phones. voice activated radio transmitters, and other applications: however. for packet voice or any speech multiplexing system in which speed and accuracy are important. their performance is inadequate.

Bell System Sound Detector for TASI Systems

The sound detection approach to bandwidth compression is not a new idea: it has been in use for many years in the commercial telephone network. Many specific implementations have been developed for this purpose.

One such sound detection technique used was engineered in the early 1960s by the Bell System for use in the Time Assignment Speech Interpolation (TASI) system of trunk usage optimization [1. 6]. This system is divided into two major components: a speech detector and a computer analysis program.

The speech detector consists of a level detection device and a flip-flop with its reset line clocked at 200 Hz. If the sound threshold is exceeded during the interval between resets. the flip-flop is set indicating sound detected within that frame: otherwise. the flip-flop will remain in the off state indicating that nothing was "heard." The resulting sound/silence status information is then passed to the computer program to smooth out the 'spurt-gap' pattern. eliminating intersyllable gaps and thus avoiding chopping off the first parts of words. caused by switching into silence just as a word is beginning.

Each spurt or gap has a duration of an integral multiple of the 5 msec frame time. All spurts shorter than some predetermined throw-away time are discarded to eliminate spurious background noise. Gaps in the remaining stream that are shorter than a given fill-in time are ignored and considered to be part of a talkspurt.

With this technique. three parameters must be adjusted: threshold for the speech detector. fill-in time, and throw-away time.

1. Threshold level. above which signals are considered sound and below which they are silence. was selected by investigating data from analysis of real speech and by trial and error. If this level were too high. syllables would frequently be clipped: if it were too low. the "TASI-advantage" would be significantly reduced.

2. Fill-in time is the maximum length of a silence period (gap) to be ignored. If sound were present. stopped for a time less than or equal to the fill-in time. and then resumed, no silence would be recognized. If, however, the silence period exceeded the fill-in time. the entire period would be considered silent. If the fill-in time were too long, valuable bandwidth would be wasted; if it were too short, syllables would be lost.

3. Throw-away time prevents short spikes of noise from tripping the detector. This is the maximum length of a burst of sound that can be totally ignored. The sole function of throw-away time is to conserve bandwidth. In the absence of this parameter, the user would not be noticeably affected. The bandwidth usage, however, would increase substantially.

The Bell System method has been tested, its parameters have been fine-tuned, and it is now in wide use throughout the telephone network. The scheme is fast and reliable--a definite improvement over VOX-type detectors. Due to its fixed threshold, however. it does not adapt to variable background noise conditions; this rigidity can cause less than optimal bandwidth usage in medium- to high-noise environments. Also, excessively quiet sounds such as whispering will be degraded.

A Dynamic Sound Detector

Human speech. due to its nonstationary characteristics. is easily distinguished from the relatively stationary pattern of background noise. Noise tends to have variations in amplitude significantly smaller than those of speech. occurring relatively slowly. By extending the same concepts behind the fixed threshold detector. dynamic sound detectors [3] can therefore be constructed which *learn* the ambient background noise level and vary the threshold. continuously adapting it for optimum noise reduction.

By analysis of short-term variations in the amplitude of the channel signal. noise is differentiated from speech patterns. Continuous averaging of the noise levels over a short time provides a reasonable basis for adjustment of the sound threshold. Fill-in and throw-away times as applied in the TASI sound detector are used to further smooth out the pattern, providing an accurate indication of the sound/silence status. Such a dynamic detector, which can closely follow the actual speech patterns. is ideal for the packet voice application.

STNI Dynamic Sound Detector

The goal for the STNI application was to implement a sound detector that would adapt to the ambient noise environment, react sufficiently quickly so as not to clip syllables. and decay quickly, to conserve as much of the available bandwidth as possible. In addition, it was desired that the sound detection be accomplished without adding to the hardware complexity.

The result was a completely software-driven dynamic-threshold sound detection system, developed specifically for the STNI. It can directly analyze raw $\mu = 255$ PCM data, detect sound. and control the speech flow in real time, within the limitations of a Z80 microprocessor.

Once per sample:

```
avg = abs(x)/256 + (255*avg)/256;
```

Once per parcel:

```
thresh = minavg + 8;
if (avg > thresh)
        {sound = TRUE;
         count = count - 1;
         if (count == 0)
                {minavg = minavg + 1;
                 count = 16;
                 }
        }
else    {sound = FALSE;
         if (avg < minavg)
                {minavg = avg;
                 count = 16;
                 }
        }
```

Figure 4: STNI software sound detection algorithm

The STNI sound detection algorithm is shown in Figure 4. The algorithm consists of two paths: the averaging of PCM amplitude values (avg), performed each sample time, and the threshold adaptation and sound/silence decision, executed at the end of each parcel.

In this algorithm, the threshold (thresh) always floats at 8 units above the minimum average (minavg). Parcels averaging in excess of thresh are passed as sound; otherwise they are flagged as silence and are not sent over the packet network. When the average amplitude level (avg) remains above thresh for 16 parcel times, minavg is incremented, thus increasing thresh by one as well. Whenever avg drops below minavg, minavg is immediately set equal to avg, adjusting for sudden drops in the amplitude. Figure 5 shows the performance of this detector on voice signals and background noise.

The threshold requires a relatively long time to climb high enough to block out a loud background noise level. The sudden absence of the loud noise, however, will be compensated for immediately. This effect is shown in Figure 6.

Speech is characteristically perforated by momentary periods of silence. Therefore, the threshold should never have time to climb above the level of the speech. Signals other than speech may not survive as well. Continuous sounds such as a dial tone or the carrier of a dataset, for example, will eventually be blocked out completely. As a result, voice-band data modulation will not function over this system. This does not present a real problem since the packet-switched digital network can handle high-bandwidth data communication directly, without voice-band modulation.

This sound detector is fast enough to detect silence between words and sometimes even between syllables. It does not, however, clip utterances either at the beginning or the end of the talkspurt. Its dynamic threshold allows it to adapt quickly to the background noise level. The sound from an extremely noisy environment, such as a noisy computer room, can make the sound detector slightly sluggish because of

8
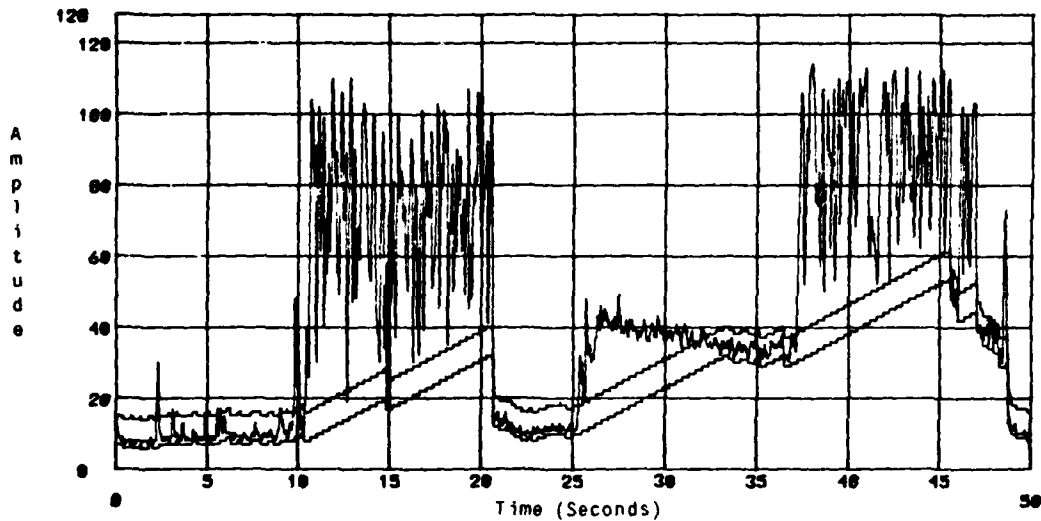
In this graph, the lower of the parallel lines corresponds to minavg in the algorithm. the higher line is thresh. The leftmost segment consists of background room noise and what little sound seeped through the wall from an adjacent computer room. Tne second segment is speech with the same background noise level. In the third segment, the door to the adjacent computer room was opened to provide a loud background noise. Note that the silence threshold climbed to eclipse this noise level completely. In the fourth segment, speech was again recorded. this time with the computer room background noise. demonstrating that the detector is still quite effective with the increased noise level. The last spike toward the end was produced by the slamming of the computer room door. Note the sudden decrease in the threshold level immediately following that event.

Figure 5: STNI sound detector performance with speech and background noise

the nonlinear response of the PCM conversion, but the sound detector still performs reasonably. and the background noise is completely eliminated.

ECHO SUPPRESSION

Echo is a serious problem with long-distance communications. Echo problems are a result of the conversion between the two-wire local subscriber loop to the four-wire telephone handset. In long-distance connections, additional conversions are done from the subscriber loop two-wire circuit to the long-distance carrier, which is typically a four-wire connection, and back to the subscriber loop on the distant end. In a local connection. the time between the initial speech and the echo is only a few microseconds, sufficiently short that it is not apparent. When the connection spans several thousand miles, however, the transmission delays are measured in milliseconds--long enough to be sensed by the human ear. When the signals are transmitted via satellite. the 44,000-mile trip takes about a quarter second at light speed in addition to any delays introduced by the transmission equipment itself.

In the commercial long-distance networks. loss over long analog transmission lines helps reduce echo since the voice travels from one end to the other. but the echo is transmitted once from the local station to the distant receiver and then back, doubling its effective loss. In digital systems. however. there is no inherent loss, and the echo becomes more apparent. To take advantage of this effect, about 6 dB of loss
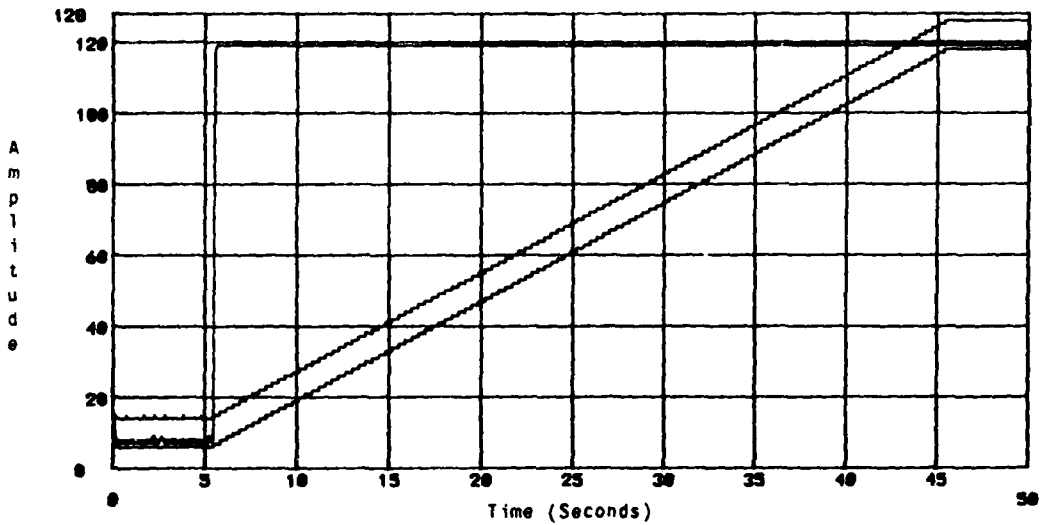
Shown here is the response of the sound detector to a constant, loud tone, in this case a touch-tone. The threshold climbs slowly, eventually passing the level of the loud tone. Once it has reached this point, it ceases to climb, stabilizing at a level just above the amplitude of the tone. When the tone was removed (not shown here), the threshold dropped immediately back to its level at the beginning of the graph.

Figure 6: STNI sound detector performance with high-level tone

has been introduced into the analog interface of the STNI, resulting in a round-trip echo loss of about 12 dB, as shown in Figure 7. This loss helps reduce the problem, but is not sufficient, so an echo suppression or cancellation device must also be employed.
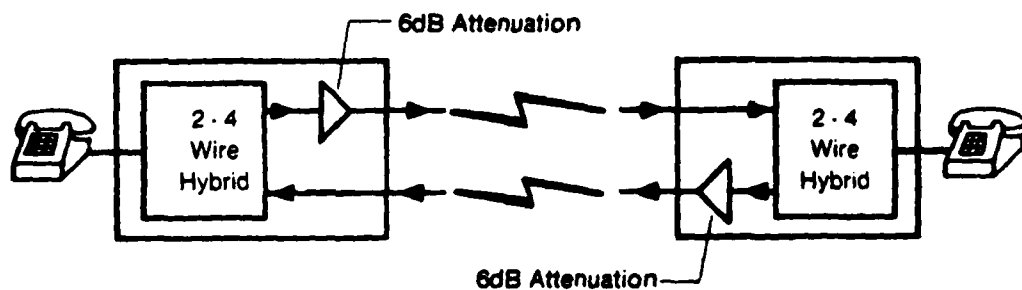

Figure 7: Loss introduced in STNI analog interface

Echo suppression attempts to block echo by blocking outgoing signals when incoming signals are received. Each end of a connection performs this function in an

effort to prevent echo from returning. Some echo passes in the transition from sound to silence, as the echo suppressor turns off. The major problem with this technique. however, is its inability to deal with both ends generating signals simultaneously. When both parties on the connection transmit concurrently. neither will be heard. This often results in choppy speech if one party attempts to interrupt the other. In a system which has very long delays. typical of satellite transmission, both parties often try to speak at the same time. Neither party is aware of this until the long delay has elapsed. at which point both stop talking and each awaits the other. repeating the process until one waits and the other does not.

In contrast. echo cancellation [8] allows full-duplex traffic, providing a much more comfortable environment for conversation. An adaptive echo canceller automatically synthesizes a filter with the characteristics of the echo path, processes the incoming speech using that filter, and subtracts the resulting estimated echo from the actual outgoing signal. As a result, the echo is eliminated from the signal.

Commercially available echo-cancellation devices are expensive and often larger than the entire STNI card. The possibility of using a digital signal processing chip to perform echo cancellation is being investigated as part of the project: however, until this work is completed, an echo-suppression algorithm integrated with the sound detector is in use. making the connection essentially half-duplex. not unlike long-distance trunks in the commercial telephone network [4]. When the party at one end of a conversation is speaking into the system. the other side is temporarily blocked from transmitting. Loud sounds override this mechanism. so it is possible to interrupt the speaking party by speaking loudly. The echo suppression is accomplished by increasing the minimum sound threshold for transmission when data is being received.

DISCONNECT DETECTION

At the end of a telephone call, when the caller hangs up. the connection over the telephone network is broken. If a call is in progress via the STNI card when this happens, this call should be disconnected and the STNI made available for another call. On the surface, this problem may appear simple to handle, but due to the idiosyncrasies of the various telephone switching systems. it is rather difficult.

Several common methods exist for detecting when the calling telephone line disconnects. Computer dial-in modems must detect a disconnect in order to relinquish the line when the user hangs up. In a modem. however, carriers are always present during a connection, making disconnection detection a simple matter of noticing the absence of this carrier tone. The STNI, however. does not have a continuous carrier, and therefore must detect other indications.

One method is to use line voltage monitoring. In most modern central office switching systems, the line voltage momentarily drops to zero when any significant event, such as disconnection, occurs. A detector need merely signal the processor when this zero-battery condition is sensed. Some older switching systems, and many modern digital PBX systems, however, do not provide this signal and thus would not be compatible with this detection method.

Another method is to listen for a dial tone. After a call is disconnected, almost all switching systems will eventually return to a dial tone to allow another call to be placed. Dial tones vary in frequency and amplitude from system to system, making absolute dial tone detection somewhat difficult.

Analysis of several dial tones from several different sources, including Bell System #1A ESS and GTE EAX central office switches and a Stromberg Carlson Crossreed PBX system, has shown that dial tones share a common characteristic in that they maintain a constant amplitude for a period of several seconds. Most systems, however, stop playing the dial tone after a period of time and revert to a loud error tone, in case the user accidentally left the phone off the hook. The dial tone on the Stromberg Carlson Crossreed PBX system was by far the shortest, lasting only about eight seconds.
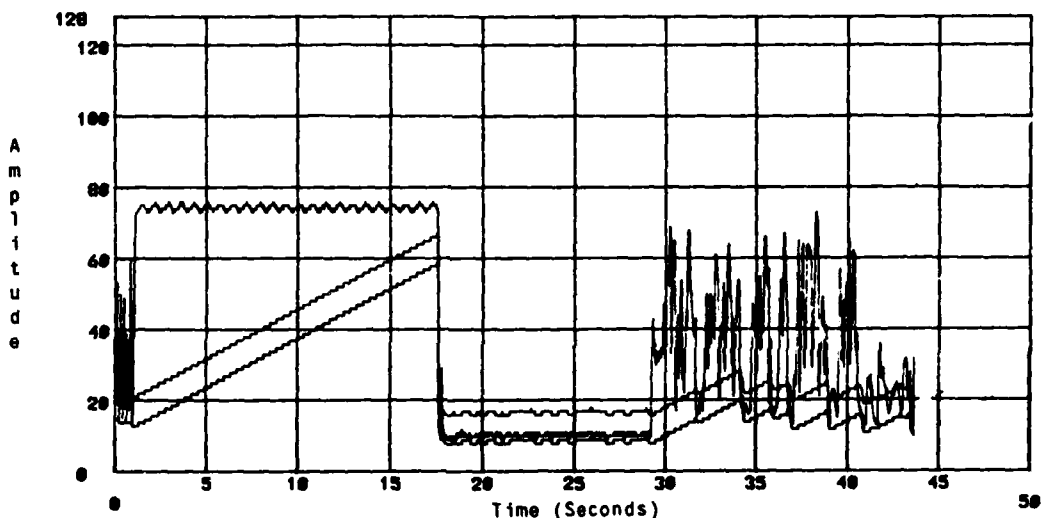

This is the graphic display of the #1A ESS dial tone, via a foreign exchange repeater circuit and roughly 5 to 6 miles of wire. It is shown here followed by a period of silence and the off-hook "scream" tone. This was the longest of the dial tones observed: about 16 seconds.

Figure 8: Bell System #1A ESS dial tone

The method chosen for use in the STNI is a software analysis of the data flowing from the telephone side of the interface. It is assumed that it is highly unlikely for the signal amplitude to remain stable for more than a second or two during a normal conversation. Figure 5 is a representative sample of amplitude variation in normal speech. The use of data modems is already precluded by the sound/silence detection method, so this assumption poses no further restriction.

Eight seconds is more than long enough to recognize the stability of a tone. Graphic representations of the collected dial tones (Figures 8, 9, and 10) indicated that the average amplitude--a statistic already maintained by the STNI--varied no more than about ±5 PCM sampling units during the time they were played. Using this information, an algorithm was developed that tracks the average amplitudes over a period of time and senses relative stability. Eight seconds was chosen as the validation period, since this is the minimum length of any dial tone studied.

12
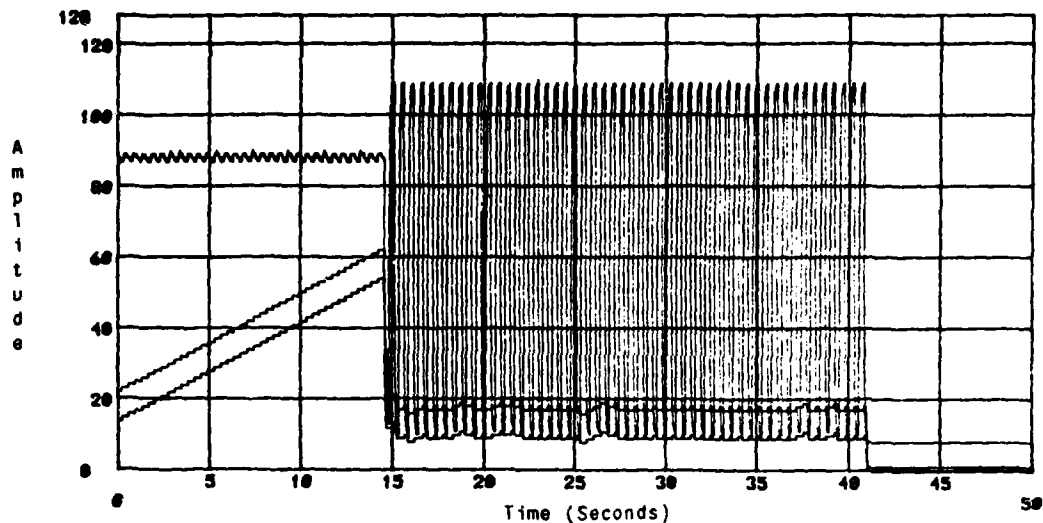
Shown here is the dial tone collected from a GTE EAX switching system. This was collected from a short tail of no more than a few thousand feet. This particular switch does not provide any zero-battery signalling after its dial tone times out, but instead switches directly to a reorder tone which lasts indefinitely.

Figure 9: GTE EAX system dial tone
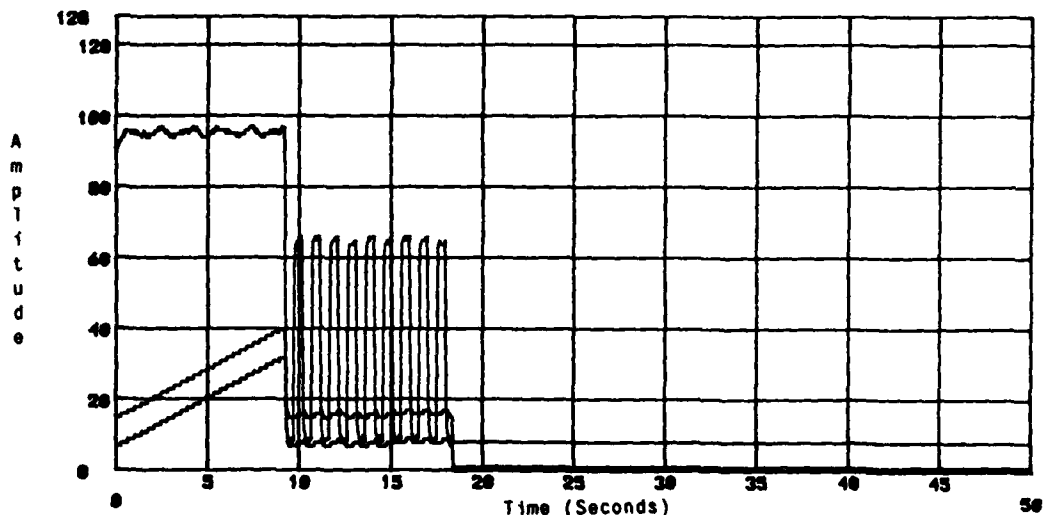
This sample was collected from a Stromberg Carlson Crossreed PBX system. This was by far the shortest dial tone collected, lasting only about eight seconds, then terminating in an indefinite reorder tone without any battery signalling.

Figure 10: Stromberg Carlson Crossreed PBX system dial tone

To prevent false trips caused by stable background noise, a minimum level is needed, below which sounds will not be interpreted as dial tones. All dial tones studied were in excess of 64 PCM sampling units of amplitude. Therefore, the detector has been set up to terminate a call on any tone with stable amplitude above the level of 64 PCM sampling units for eight seconds.

Some switching systems do not revert to a dial tone at the end of a call, rendering the dial tone detector ineffective. Two other signals are common: silence for an indefinite period, or a pulsed tone also for an indefinite time. One solution to the problem would be to define "absolute silence" and devise a detector for it. This would handle the bulk of the systems that don't revert to dial tone. In the next major release of the STNI software, this function will be incorporated. Pulsed tone detection, however, is somewhat more difficult, and may not be necessary. One other safeguard is to apply a timeout to any state other than an active conversation. This is really a function for the device controlling the STNI, since it administers the actual user interface.

DTMF DIALING

Software-generated DTMF signals are used to dial out on the telephone line. These tones are played from prestored sections of ROM, each containing one or more complete envelopes of modulated tone as a sequence of PCM samples. The dual frequencies were generated and mixed in advance by a simple FORTRAN program and then placed in the STNI ROM. The length of each table was chosen by the FORTRAN program to be a large enough multiple of the wavelength so that the resulting frequency would be within $\pm 0.5\%$ of the target frequency. The DTMF specification allows for an error of $\pm 1.5\%$. This mechanism is also used to generate industry standard dial, ring indicate, and busy/reorder tones for familiar user feedback signals. However, frequency tolerances for these tones are somewhat relaxed to save ROM space. Table 1 is a list of the frequency pairs used for each tone.

Initially, the dialer was made nearly as fast as the industry standard specification provides: roughly 50 msec on, 50 msec off. The Bell system # 1A ESS was able to handle full-speed DTMF signalling. Some systems, however, could not parse input this fast. The GTE EAX was only able to handle tones with about a 70 msec on and off time. As a result, the interdigit delays were increased to accommodate the slower switching equipment.

SPEED

The actual implementation of the software is not in the C language, as pictured in the example in Figure 4, but in Z80 assembly language. Compiled code would not have provided sufficient speed to handle the detection and tone generation functions in real time.

The Z80 microprocessor is clocked at 3.072 MHz, from a 6.144 MHz crystal that also provides clock signals for the PCM CODEC and other circuits. The microprocessor receives an interrupt from the CODEC 8000 times per second, indicating a data-ready condition. The interrupt code handles data transfers, sound detection, and stored tone generation. A total of 384 instruction cycles are available to the Z80 between these interrupts, not all of which may be used by the interrupt code, since there is also a control process that requires a small percentage of the available CPU time.

Table 1: Tones generated by the STNI

Tone	Target frequency	Actual frequency
"1"	697Hz x 1209Hz	698.4127Hz x 1206.3492Hz
"2"	697Hz x 1336Hz	698.4127Hz x 1333.3333Hz
"3"	697Hz x 1477Hz	695.6522Hz x 1478.2609Hz
"4"	770Hz x 1209Hz	767.1233Hz x 1205.4794Hz
"5"	770Hz x 1336Hz	771.9298Hz x 1333.3333Hz
"6"	770Hz x 1477Hz	771.9298Hz x 1473.6842Hz
"7"	852Hz x 1209Hz	848.4848Hz x 1212.1212Hz
"8"	852Hz x 1336Hz	848.4848Hz x 1333.3333Hz
"9"	852Hz x 1477Hz	854.3689Hz x 1475.7281Hz
"*"	941Hz x 1209Hz	941.1765Hz x 1210.0840Hz
"0"	941Hz x 1336Hz	941.1765Hz x 1333.3333Hz
"#"	941Hz x 1477Hz	941.1765Hz x 1478.9916Hz
Dial	350Hz x 440Hz	351.6483Hz x 439.5604Hz
Ring	440Hz x 480Hz	436.3636Hz x 484.8485Hz
Busy	480Hz x 620Hz	486.9565Hz x 626.0870Hz

The current implementation consumes about 82 percent of the available time when transferring PCM signals. leaving only the remaining 18 percent for the control process. Linear predictive coding (LPC) transmissions require slightly less of the processor, and tone generation, still less. It would be possible to increase the processor clock speed to add further capabilities, if necessary. although additional circuitry would be required.

SUMMARY

A compact, intelligent telephone interface has been developed. based upon a Z80 microprocessor, providing a connection between a commercial telephone line and a Lincoln Laboratory Packet Voice Terminal for use in large-scale packet voice experiments. Commanded by an ASCII serial line, the device implements sound/silence detection, tone signal generation, and telephone line control, providing an extremely flexible interface that allows calls to be placed between the commercial switched telephone network and the DARPA Wideband Packet Satellite network.

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