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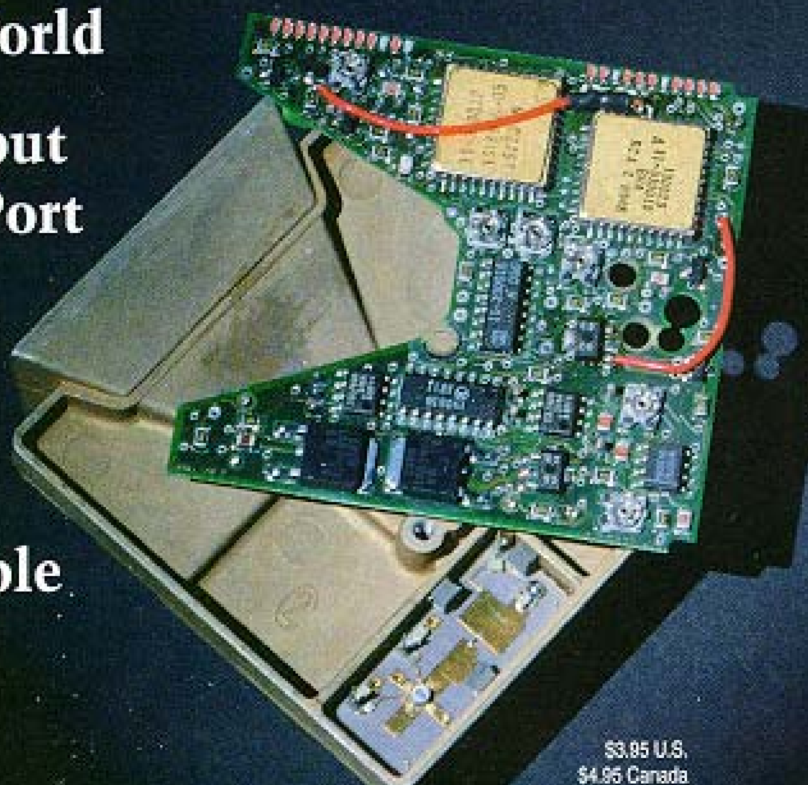

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Connect the Personal PBX to the Real World

FEATURE ARTICLE

Richard Newman

Richard presents a circuit that connects the Personal PBX small switch with the "outside" world—the telephone company central office. A SLIC interface used in conjunction with the PBX enhances conversation quality.

In *INK* 48, I addressed everything you need to call from one telephone to another, including dial tone, DTMF (Dual Tone MultiFrequency) decoding, ringing the called party, and setting up a conversation through an analog-switch matrix.

In this article, I want to take you outside your own phone system. I present a circuit that interfaces the Personal PBX, a small switching system, to the outside world so that a telephone company central-office interface may be connected.

I'll also present an improved subscriber-line-interface circuit (SLIC) which, when coupled with the central office (CO) interface, reduces the amplitude and frequency loss of the conversation.

SLICs AND COs

Let's review the difference between a SLIC and CO interface. The SLIC provides loop current to the telephone set. Ringing signals indicate a call has arrived.

The CO interface terminates the loop current provided by the central office and detects a ringing condition applied to the line. These definitions apply to the system you are building or working on. A CO interface can:

- detect a call sent from the central office
- terminate the line answering or originating a call
- interface the central office with the speech matrix transfering the audio from one 2-wire circuit to another
- detect line current indicating if the caller hung up before we have
- protect itself from lightning and overvoltage surges by breaking the physical connection when a failure occurs

Figure 1 shows the line interface. As you can see, it includes metallic coupling, loop-current sensor, polarity reversal correction, and the line seizure relay.

When relay K1 is not activated, DC loop current does not flow. The circuit is considered "on hook" or ready to receive an incoming telephone call.

When the central office applies 90 VAC at 20 Hz to ring the line, AC current passes through the nonpolarized coupling capacitor C1. This AC current flows through the optoisolator. Its output at the not-signal point is a 20-Hz square wave which is TTL compatible.

The not-signal output is fed to the system's microprocessor. Its frequency and duration reflect the central office's ringing pattern. Personalized ringing, available to electronic central offices, enables up to three telephone numbers to signal one physical telephone line. The pattern of the ringing determines which logical line is signaled. Through software which decodes this pattern, the switch determines the personalized ring signaled by the central office.

Because the transformer T1 is in the circuit when incoming ringing voltage is applied, the circuits attached to the transformer's secondary must be protected. Zener diodes D1 and D2 clamp the waveform to whatever voltage you choose. For example, a 12-V zener limits the positive and negative swings to around 12 V.

When K1 is activated, loop current from the central office flows through the optoisolator and causes it to light continuously. The opto's not-signal output goes low and stays low for as long as loop current is flowing.

Any time the central office sends a supervision pulse-which appears as a momentary loss of loop current-the opto's output momentarily goes high. A supervision pulse is sent from the CO when the calling party hangs up before you do.

Current flowing through T1 causes the audio signal on the CO to couple with the 2-wire analog port and vice versa.

Diode D3 protects the opto during spikes when the circuit is seized. Fuses F1 and F2 protect both the interface and central office from failure or high-voltage surges. When a surge hits tip, ring, or both, one of the MOVs sinks the current to earth ground, and the associated fuse opens. When the fuse opens, arcing occurs if the current is sufficient, so the MOVs need to be of sufficient rating to handle it.

Resistors R1 and R2 may not be necessary in everyone's opinion, but they provide a small drop across them. This drop reduces the amount of cur-

SIGNAL CONVERSION

Figure 2 shows the 2-wire-to-2-wire converter which interfaces the 2-wire port of the line interface to the 2-wire port of the analog-switching matrix.

The 2-to-2-wire converter consists of two back-to-back 2-to-4-wire converters. The gain in any one direction is always less than one. However, the complexity of the implementation depends on the terminating impedance of the 2-wire side as well as the transmission characteristics of the CO line, SLIC, and switch matrix.

Looking at the right half of Figure 2, you can see that a typical 2-to-4-wire converter is made up of two op-amps

ceive signal has a portion of the transmitted signal with it and you try to convert it to 2-wire, the converters form a closed loop and oscillate.

If you reduce the gain, you eliminate the oscillation, but your goal of reducing the loss by adding gain won't be accomplished. You might as well have connected the coupling transformer directly to the switch matrix.

The measure of the converter's ability to separate the transmit and receive signals is called *transhybrid rejection.* Depending on the application, there are several ways to improve rejection. A CO interface might be more reactive with elements of resis-

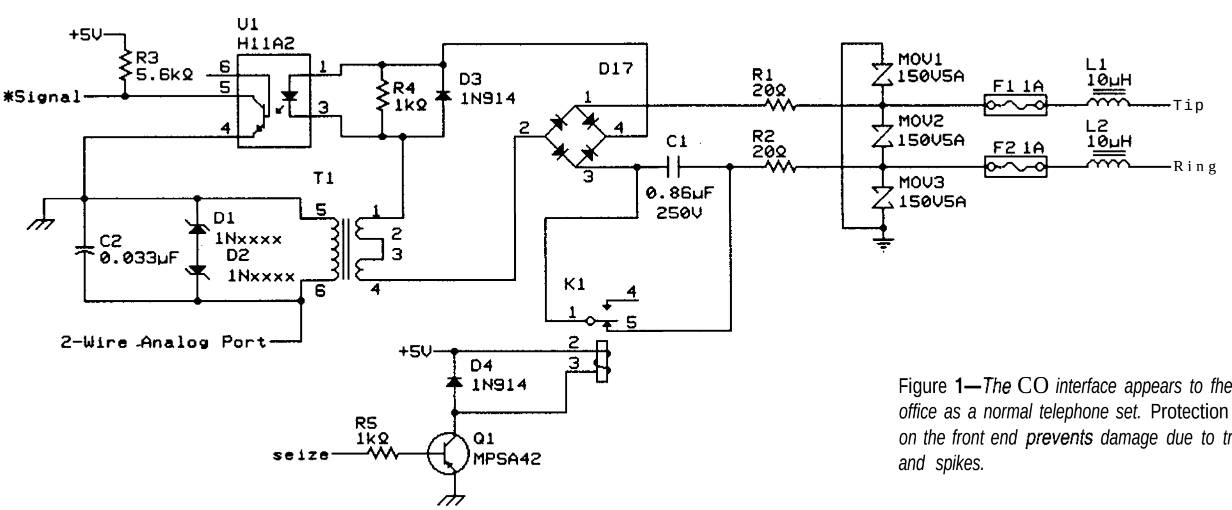

Figure 1—*The* CO *interface appears to the central office as a normal telephone set.* Protection *circuitry on the front end* prevents *damage due to transients and spikes.*

rent across T1 so a wider variety of loops can be accommodated. These resistors are required for real-world performance.

Diode bridge D1/D2 permits the CO's connection to the interface regardless of its polarity or whether the CO inflicts polarity reversals on the circuit. Again, real-world situations require that anyone be able to hook up a telephone line. The bridge ensures that hook up occurs even if the wires are reversed.

Relay K1 is activated by a simple inverting driver. A TTL low on the seize input causes the interface to take the telephone line off hook and draw a dial tone from the CO. A high on the seize input causes the interface to drop the call (if one was in effect) and return the telephone line to the idle or on-hook state.

and a handful of discrete components. This converter transmits the input signal from the 4-wire input side to the 2-wire bidirectional side but keeps the signal from showing up on the 4-wire output side. It transmits to the 4-wire output any signal which shows up at the 2-wire bidirectional side but not on the 4-wire input.

Since the converter separates a transmit signal from a receive signal, you can add some gain to the output of the converter to overcome the insertion loss imposed by the physical-line interface. The amount of gain you add depends on how well the converter separates the signals. If it can't do the job well, the transmit signal appears in the receive signal.

While this mixing of signals might be acceptable for some applications, it's disastrous for ours. When the re-

tance and capacitance, but a very short SLIC might be almost purely resistive.

Resistor R6 is 680 Ω because most central office lines lie somewhere in the range of 600-900 Ω. Note that the resistor's value matches the attached line's impedance, not the resistance of the transformer's secondary.

Resistor R21 is the terminating impedance for the analog-switch matrix. Because R21 faces another resistor of the same value (R21 in the other circuit), when two 2-to-2-wire converters are in a conversation through the switch matrix, the reflected energy is minimized and rejection should be excellent.

Continue looking at the right side of Figure 2. The signal applied to U2a pin 3 appears at the 2-wire analog port but not at U2c pin 7. The input signal is subtracted from the output signal.

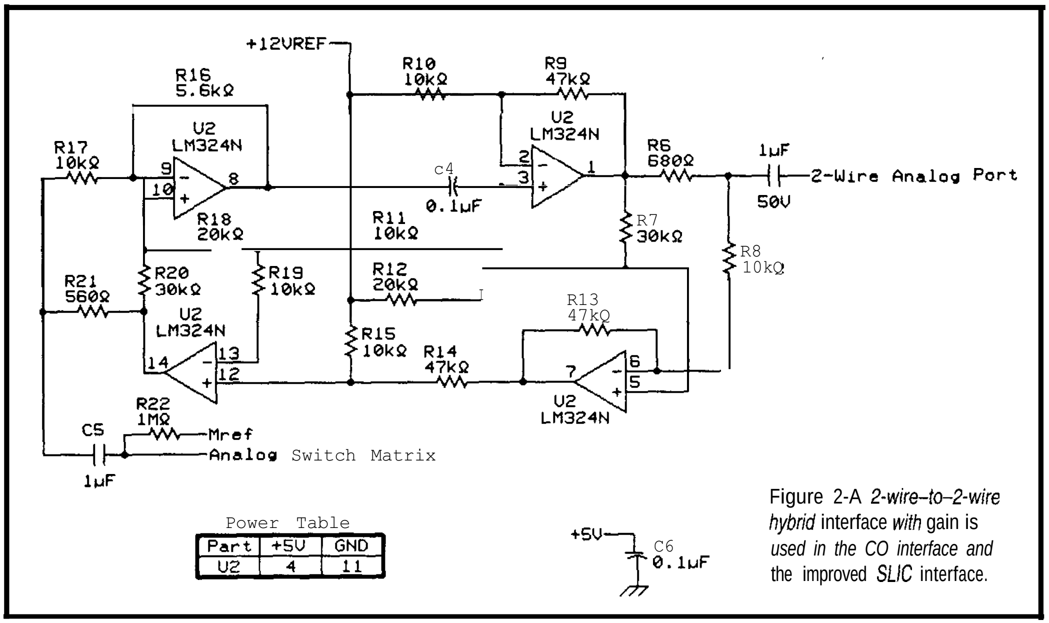

Figure 2-A *2-wire–to–2-wire hybrid* interface *with* gain is *used in the CO interface and* the improved *SLIC* interface.

You therefore minimize loss of the signal injected into the 2-wire analog port and transferred to the balanced subscriber line. The small loss imposed by the switch matrix and capacitive coupling can be reduced further by the gain added in the 2--to-2-wire converter.

A typical call, shown in Figure 4, arrives at the phone line in 2-wire format. It terminates when the CO interface activates. The audio signal is coupled to the interface's 2-wire output, and from there, it is passed to the first 2--to-2-wire converter. Inside the converter, it is made 4-wire by separating the transmit and receive portions.

The separate signals receive added gain in the process. They are then recombined into another 2-wire signal, which is sent to the analog-switch matrix. This matrix passes the signal to the selected output where the process is reversed, only now the signal is sent to an extension interface. The result is a quality, full-duplex conversation between a telephone extension on the PBX and the outside phone line.

The lefthand side operates identically.

Resistor R22 provides the DC carrier for the AC signal to ride through the single-supply switch matrix. The DC carrier is less than half the switch-matrix supply voltage.

Note that the Personal PBX is a positive-supply-only system. Thus, all of the analog portions have a reference supply. In this case, reference is 12 V.

In my last article, I described a SLIC that had a transformer in it with a purely resistive feed. While this works well, we can do better.

THE **SLIC** INTERFACE

Figure 3 shows the SLIC which must be used with the converter illustrated in Figure 2. The features of this model include:

- reduced parts count
- elimination of the transformer
- a constant current feed that gives subscriber-line low-impedance characteristics to DC and high-impedance characteristics to AC

The SLIC shares the same relay drive and current detector as the CO interface, so controling the SLIC is uniform in the system. Making stalring low deactivates relay K1, so the telephone set is attached to the audio interface. When stalring is high, tip is removed from the audio interface and applied to a common ringing bus so that the telephone's bell can operate.

When the phone attached to tip and ring is lifted, current flows through R5 and R6 (both 20-Ω resistors). Current is limited to a rate set by the ratio of R2 to R4 and the resulting voltage (measured at the base of the PNP transistor Q2}. When current flows, the optoisolator lights, signaling the CPU that the telephone set has been lifted off hook.

The act of limiting the set's current to one specific flow rate makes the subscriber-line low-impedance to DC signals (the 24-VDC carrier which powers the telephone set) and high-impedance to AC signals (the speech which rides on top of the DC carrier).

SOFTWARE CONTROL ISSUES

If you downloaded the code for the original project from the BBS, you'll notice that the system CPU must keep

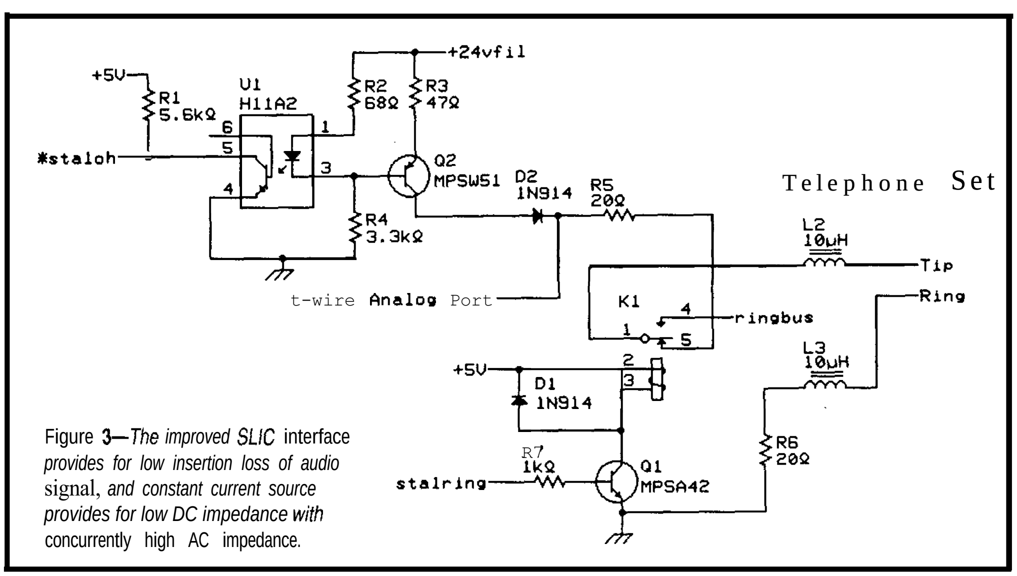

Figure **3**—*The improved SLIC* interface *provides for low insertion loss of audio* signal, *and constant current source provides for low DC impedance with* concurrently high AC impedance.

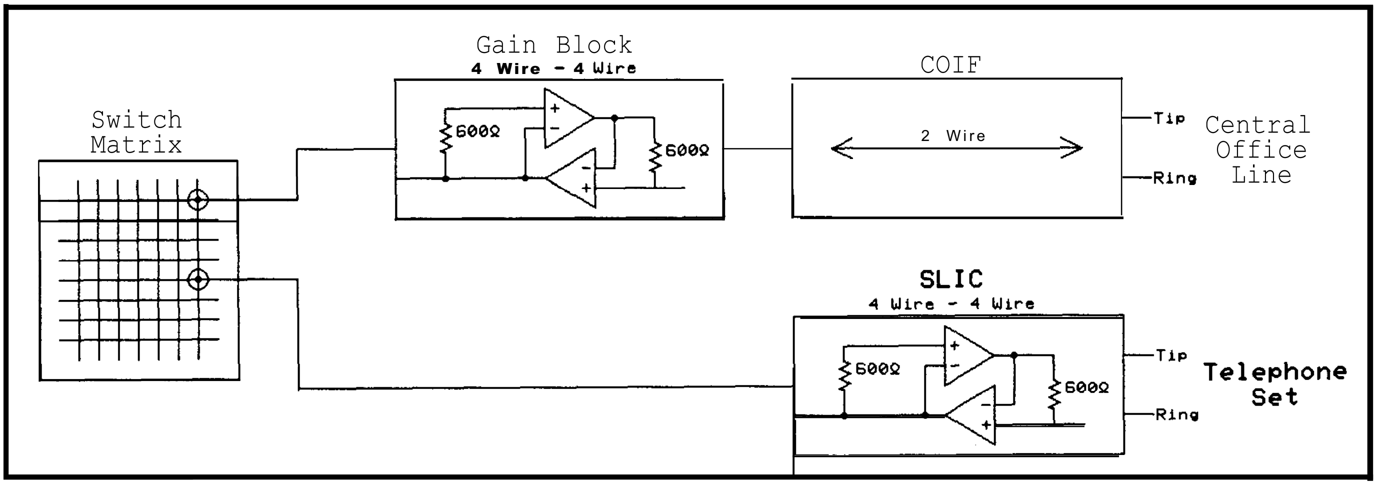

Figure 4— A conversation goes through a number of building blocks to get from the CO to the telephone set.

track of finite and shared resources. One DTMF receiver, for example, can service all the phones on the system, albeit one at a time.

Once the finite resource is used and no longer needed, it must be freed up, even if the call established is still in progress. This same resource management and queuing must be applied to the central-office circuit, touch-tone sender, and call-progress receiver.

Since you're adding central-office-calling capability to your switch, you must now make a decision about how external calls are to be set up. Set up refers to the way a customer's dialed number is conveyed to the equipment connected to the central office interface.

It includes two choices. The first, called *pass through,* does not require a system to have physical DTMF sender hardware. This option saves on cost and board space.

The second choice is called either store *and forward* or *selective/adaptive routing.* This option requires a DTMF generator to be available to the system.

If you choose pass-through routing, you can implement features such as toll control. However, you can't selectively route the call to a preferred destination to implement features like least-cost routing or alternate operator services. These features are not possible because you don't have a DTMF sender and must depend on tones from the subscriber's line to dial out to the CO interface.

When a customer picks up a telephone set, they receive a dial tone from the PBX tone generator. The customer *must* dial a trunk-group access code, like 8 or 9. The PBX then conferences in the selected central-office circuit because the SLIC, DTMF receiver, and CO interface share a common audio path.

If you don't want toll control, you can drop the DTMF receiver at this point and attach the SLIC directly to the CO interface. As far as the PBX is concerned, as soon as the trunk-group access code is dialed and the user receives the outside dial tone, the call is set up and in progress. No common resources are assigned, and the call's next state is to hang up.

If you choose the pass-through arrangement and want toll control, you must monitor the digits the customer dials to the outside line. If the customer dials digits conflicting with a preferred dialing pattern, the CPU releases the outside line and gives a busy signal from the PBX's dial-tone generator.

Once the toll-control plan is satisfied (i.e., enough digits are dialed that the system knows where the call is going), it releases the DTMF receiver even if a complete number is not dialed. After all, if the customer dials anything other than 0 or 1 for the first digit, you can be sure it won't be a long-distance call. There's no need to keep the DTMF receiver assigned?

There's one exception to this rule. To provide station message-detail recording (a record of the date, time, trunk, station, and number dialed) to the serial port when the call hangs up, keep the DTMF receiver assigned long enough to get the complete number.

STORE AND FORWARD

If you implement either the store-and-forward or adaptive/selective-routing algorithm, you'll be rewarded with superior flexibility in handling a call. When users dial, they dial into the PBX. The central processor stores dialed numbers in a queue and analyzes them to determine the most efficient and inexpensive way to send a call through the public network.

As the user dials, the central processor sets up a concurrent call within the system between the CO interface, a DTMF sender, and a call-progress receiver. This call has its own speech path, so the user doesn't know a concurrent call is in progress and can't hear the interaction between the CO and CPU setting up the outgoing call.

Using store and forward, the CPU can insert numbers into the user's dialed digits. It can even call an alternate long-distance carrier, wait for a second dial tone, and dial the customer's long-distance account code plus the number that the user originally dialed from the calling station.

Again, the network-access-control activity is going on behind the scene without the user's knowledge. The

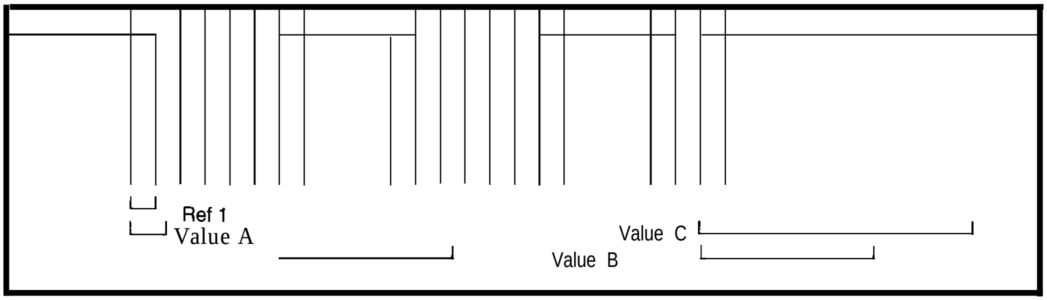

Figure C-Using three counters and a timer, it's possible to *detect the presence of a ring signal, the cadence (for distinctive ringing), and the number of rings.*

store-and-forward method is useful if you want to make the PBX as transparent as possible and yet wish to provide sophisticated features.

INCOMING CALLS

Of course, incoming calls must also be considered. You may want to provide configurable options in the computer's memory dictating which port to forward a call to when it arrives on a CO circuit.

For example, you might send a call to station 10 when it arrives on CO1, while another call goes to station 15 when it arrives on C02. You may also have calls forward to another extension after the call has rung the first line for a selected number of rings.

You can easily add unique features by changing the supporting software (rather than hardware). For instance, you can decode the personalized ring from the telephone company and send the call to a different station for each of the personal rings. Thus, one phone line in a home could receive calls for up to three people and automatically route the call to the appropriate extension and answering machine.

To implement a feature like personalized ring decoding, you need to monitor the I/O port connected to the CO interface's optoisolator. When the optoisolator detects ringing from the central office, the associated I/O port toggles between 0 and 1 at the ringing frequency.

Figure 5 shows the digital output of the optoisolator. The algorithm goes something like:

1) when low, reset counter 1 to 0
2) wait for high
3) when high continuously, increment counter 1
4) if counter 1 exceeds preset value A, CO ringing is no longer present. Increment counter 2 (i.e., the number of bursts being sent).
5) if low, return to state 1
6) if signal stays high and counter 1 exceeds value B, increment counter 3 to count of the ringing cycles.
7) if signal stays high and counter exceeds value C, reset everything because ringing has not been present for awhile and the remote caller has abandoned the incoming call. Reset the station being rung to idle.

Value A is longer than one cycle of the frequency the central office rings. If the variable is larger than this value, central office is no longer signaling.

Value B is longer than the longest burst. If ringing is encountered again before the variable reaches this value, you're still in the ringing cycle but have encountered another burst.

Value C is longer than the longest complete ring, inclusive of all bursts. This value helps determine when an incoming call is abandoned.

Of course, when an incoming call is detected, you ring the destination station and let the CO interface remain idle. The caller receives ringback from the central office, which sees the call is not yet answered. When the destination station answers, engage the CO interface, making an audio connection between the CO and SLIC.

If you answer the CO interface and provide ringback from the PBX, you consume finite resources (i.e., call progress sender and the DTMF receiver). Then, if the destination station doesn't answer and the call is long distance, the caller is charged because the CO interface was activated and answered.

TIME TO HANG UP

The interface presented here improved conversation quality dramatically, making the Personal PBX a full-fledged switch rather than a super intercom.

These same interfaces can be massaged into a wide variety of systems and form factors. For instance, a board for a personal computer can provide a complete, low-cost PBX system for home and offices.

With a complete PBX, you can enhance the software for custom features such as toll restriction, remote access, and PC-based remote control of telephones.

Look for an article in April on Caller ID. I'll show you how to make a stand-alone caller ID decoder, interface Caller ID to a personal PBX, and take advantage of it in software.@

Richard Newman is an electrical engineer living in Dallas, TX. He designs specialized communications and industrial automation equipment either in partnership or on contract. He can *be reached at*

SOFTWARE

A C source code example of a call-processing algorithm implementing selective/adaptive routing is available from the Circuit Cellar BBS and on Software on Disk for this issue. See the end of "ConnecTime" for downloading and ordering information.

SOURCE

PCB, 8 SLICs, 3 COs, source code, EPROM, and schematics

Caden Development
4335 Cedar Springs, Ste. 201
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