VOIP (VOICE OVER IP)

AND THE CURRENT STATE OF TELECOMMUNICATIONS IN THE UNITED STATES

TODAY'S PHONE NETWORK

In order to explain the state of today's telecommunication network, we first must have a solid understanding of what VoIP is, what the network used to look like, and what it looks like today.

Voice over Internet Protocol (Voice over IP, VoIP) is a methodology and group of technologies for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. In simple terms, voice sounds are being converted to packets and sent across a computer network.

When voice is converted to packets, a software algorithm is used to encode and decode the sounds, which is called a CODEC. CODECs are standardized, so that the sender and the receiver know how to handle the calls.

There are many different packages of CODECs that are used depending on the needs and the available bandwidth of the network. <u>But, they all have one thing in common; the ability to vary the compression rates if necessary to keep the call going.</u>

At face value, being adaptable sounds like a good thing. But for alarm panels, it's not; because as the CODEC compresses the audio more to deal with network conditions, parts of the voice call get dropped. When compression is needed because of network congestion, CODECs start to immediately drop sounds outside the 20 Hz to 20 kHz range. CODECs also start to flatten out the audio so that things begin to sound muffled. Most healthy humans can hear a range from 20 Hz to 20 kHz so it would appear to make sense to drop sounds outside of that range. While this allows the network to flex to handle differing network conditions without greatly impacting the quality of the voice call, it has an adverse impact on alarm signal traffic being transmitted through VoIP.

In addition to the compression problems found in VoIP networks, there are also other factors that will hamper alarm transmission problems. The first is latency, usually called Jitter. This is where packets are sent spaced evenly but are not received either in the same order or not at the same pace, which causes the calls to sound like parts are cutting in and out, or as if you are underwater.

The second most common problem is out of band DTMF. DTMF/Touchtone over a standard analog circuit sends the DTMF tones as part of the overall audio, which is called in-band DTMF. Some VoIP networks, in order to conserve bandwidth, will send DTMF out of band. This means that when you press a button on a phone, or a panel is trying to send Contact ID, instead of just carrying the audio tones, a message is sent from the first CODEC to the next CODEC to turn on/off these tones.

It is also important to understand how long distance networks work now compared to 10 years ago. In the past when a long distance call was placed either directly or via Toll Free, each local carrier had "TIE" lines into the long distance provider's networks. This allowed calls to traverse the networks all on 100% analog circuits.

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Today we have gone from the network where you could hear a pin drop, to the network with the least dropped calls. However, with thousands of small local companies, it would be impossible to keep this model working; Therefore, a new industry has emerged that bridges the calls from the local providers to the long distance providers. These new companies are called Tandems or Inter-Exchange Carriers. With no exception, these carriers are 100% VoIP today, and in fact are bidding on each call in real time to be the least cost route. So it is in their best interest to compress the calls as much as possible because of the cost savings.

WHY ALARMS DON'T PLAY WELL WITH VOIP?

There are a lot of reasons, many are format specific but here are the major ones.

- With Contact ID, a DTMF (touch tone) format. If you have Jitter, latency or out of band DTMF, the most common problem is that the signal gets decoded as another format such as FBI 4X3 or SESCOA. This will typically result in an alarm being sent to the wrong account.

- The other thing that will happen to DTMF is that the turnaround time is too slow, so you will see panels sending events over and over again, because they don't hear the acknowledgement tones.

- With Modem Formats, when compression gets turned on, modem tones get left out. So even though the call is placed, the receivers cannot decode it because so much of the actual data is missing. Also, high turnaround times will cause slack problems and you will not get all of the data.

- With Pulse Formats, you will typically see missing digits resulting in improperly decoded signals. This is especially true with formats not set up for Parity but instead use double round.

WHAT CAN YOU DO ABOUT IT?

The first thing you can do is immediately stop selling alarm panels on dial-up. It is very likely that POTS lines will sunset before 2020. In reality, you cannot call anywhere today without ending up on VoIP somewhere along the line.

Start switching over your legacy dial-up accounts to other technologies such as cellular, IP or AES. It is only a matter of time before you will be forced to, so you might as well start now.

For pulse formats try to use those with Parity and avoid double round.

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For Contact ID try to isolate your CID accounts so that with DTMF formats you are ONLY using CID. Make sure that you are not also using Ademco Express, Ademco High-speed, FBI 3x4 or SESCOA for any of your accounts. If you isolate the pool to only CID, then we can set the receivers to not process any other DTMF format except correct CID with Parity. This last method will stop a lot of signals from appearing on the wrong accounts.

WHAT HAS RAPID RESPONSE DONE?

There are a number of technology based things we have done and will continue to do to help for the near term. But long term, POTS is going away.

- We only use Tier 1 voice carriers like AT&T, Level3, TW Telecom, CenturyLink etc.
- We are 100% connected to these carriers over analog TDM circuits.
- We are utilizing the most technologically advanced Avaya phone switch available in today's market.
- We consistently update receiver firmware to try and adapt to changing conditions.
- We are proactively monitoring error rates for all carriers that have alarm traffic.
- We are members of CSAA and AICC who are in contact with the FCC and the telecommunications providers.
- We have provisioned CID only profiles for Surgard receivers for those dealers that only have CID DTMF formats (other modem and pulse formats still work).
- We have special programs and equipment to help dealers get over the hump and get off of POTS without breaking the bank.