Personal Response Systems and Next Generation Networks

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VCI is a member of the Communications Alliance, (previously the ACIF), which is developing technical and equipment standards for the proposed National Broadband Network.

1. Background

Around the world, telecommunications networks are changing, from 'plain old telephone service' (POTS) networks that carry data as a secondary usage, to all-IP (Internet Protocol) data networks that carry voice as a secondary usage. The analogue 'public switched telephone network' (the PSTN) is undergoing fundamental change as it is modernised.

Broadband IP-based services employing digital subscriber line (DSL) technology are beginning to dominate the use of the copper line between customer premises and local exchanges, Asymmetric DSL (ADSL) currently being the most common. While DSL operated as an adjunct to long-standing analogue voice services for the past decade, and many Internet players began offering video as DSL speeds improved, the advent of "voice over internet protocol" (VoIP) has added yet another dimension.

Next generation networks (NGNs) will maintain only a single network for voice, data and video, enabling carriers to maximize their service offerings and increase their revenue streams. Eventually, only a single IP data connection will be provided to private customers and all customer equipment will be IP/VoIP based.

Migration to all-IP networks has been underway for some time, with trunk (inter-exchange), corporate and government networks already largely IP-based. Upgrading public networks is now under way, probably with at least one transitional stage allowing the continued operation of existing analogue customer equipment.

VoIP phones and "naked-DSL" services (lines without dial tone or analogue POTS) continue to increase in popularity but, in order to limit damage to their existing revenue streams derived from POTS, Telco's with their own POTS networks have resisted the introduction of VoIP-only phone services for as long as possible, or at least to slow down its introduction on their own networks.

Hybrid Fiber Cable (HFC) systems which support IP access networks carry video, IP data and VoIP voice services. Streaming video-on-demand may be provided by clusters of servers, possibly located at neighborhood fiber-cable nodes.

Whatever happens, the move from POTs to NGNs will affect security alarms and personal response systems (PRS), and we need to plan now for the access technologies used to connect the customer to the network, the hardware interfaces, the data protocols, and the signaling formats, to ensure a seamless migration through the various stages of network modernization.

2. The Optus Hybrid Fibre Cable (HFC) network

One example of a new generation telephone network is the VoIP based Optus Hybrid Fibre Cable (HFC) network. The Optus HFC network provides a high quality telephone service over the same cable which delivers pay TV and broadband internet services.

A cable modem is placed in the customer premises' and the internal telephone cable is disconnected from the incoming exchange line and connected instead to a "simulated PSTN" port on the modem.

The 'simulated PSTN' connection allows continued operation of standard analogue telephone equipment such as telephones, security systems, medical alarms and FAX machines etc. Battery back-up is minimal but can be extended with an optional external battery box connected to the cable modem.

Recently Optus have been converting customers from their old analogue based system to the new VoIP based system, and some security and alarm systems have failed to operate correctly.

Subsequently, a variety of medical alarms have been tested over the new generation Optus network and, apart from an incompatibility identified in a particular model of modem, there does not appear to be any difficulty connecting security or medical alarms to the Optus HFC network.

In fact, the Optus HFC network appears to be a very robust network capable of transmitting very short in-band DTMF data pulses without significant distortion. Optus are instructing their modem manufacturer to overcome the incompatibility.

On-site investigations have identified problems with ‘in-premises’ wiring after the customers services has been converted to Optus HFC. Testing the telephone connection after re-wiring may be limited to testing the basic telephone service only, and any other equipment such as security and PRS alarms may not be tested. Optus are currently working with their installation contractors to improve the outcome where alarm equipment is present.

The Optus experience has been very interesting. In many ways, it’s a forerunner of the proposed optically distributed National Broadband network (NBN). Rather than a cable modem the NBN will use an optical modem, but all other equipment and connectivity features will probably be similar.

3. The National Broadband Network, FttN, FttH, and HFTP.

In 2008, Telstra announced its intention to build a fibre-to-the-node (FttN) network, providing VDSL (very high bit rate DSL), over the existing copper lines into customers’ premises.

In April 2009 the Federal government announced plans to build a National Broadband Network (NBN) within 8 years (earlier in Tasmania) delivering very high speed access over a full fibre-to-the-home (FTTH) network.

More recently the government announced their desire to break-up the vertical integration of Telstra, separating out infrastructure from retail.

Both FttN and FttH deliver very high speed broadband by taking the fibre optic cable deeper into the network, closer to the customer.

In an FttN network the fibre optic cable terminates at some convenient point in the street, probably the pit or the post. The ‘last mile’ customer connection becomes the ‘last few hundred meters’ and much higher data speeds are possible.

Hybrid-Fibre-Twisted-Pair (HFTP) is another version of FttH where customer access to the node uses the existing Telstra owned copper twisted pair telephone cable.

In a FttN system, where each service provider is responsible for connecting their customer to the ‘node’, there could be significant difficulties for PRS/alarm providers. When Telstra was privatised it retained ownership of the network infrastructure, including the ‘last mile’ customer access, (usually buried copper). Although the ACCC attempts to regulate the price Telstra charges its competitors for wholesale access to its network, naturally Telstra has actively resisted providing its assets at less than commercial rates.

In order to avoid the necessity of using Telstra owned lines, it is very likely that in any FttN system without low cost access to the Telstra copper, a variety of ‘alternative’ access technologies would have emerged to connect the customer to the node, such as wireless (Wi-Fi and WiMAX), Broadband-over-Powerline (BPL/PLC), HFC cable, and others.

For instance, Optus connects phone customers using its HFC cable network in capital cities rather than pay Telstra wholesale access fees for use of their copper lines.

Each access technology would have its own set of technical requirements and unique customer access terminal equipment. Maybe some would be battery backed-up, maybe some wouldn't. The same goes for provision of a simulated analogue PSTN connection. In such a scenario, PRS providers would need to have a 'basket' of PRS technologies available depending on what was found at each customer's premises'.

The NBN proposal to use an FttH network, or the possibility of using a HFTP network if existing Telstra copper becomes available, promises to minimize the number of different access technologies. With FttH/HFTP, it is now more likely there will be commonality of terminal equipment located in customer's premises', allowing standardised technical requirements, and a more universal quality of service.

Although alternative access technologies may still be used in some circumstances, especially where an existing and capable access technology exists (such as HFC cable), VCI believes the decision to adopt FttH/HFPT network architectures greatly simplifies the provision of future PRS.

VCI is a member of the Communications Alliance, (previously the ACIF), which is developing technical and equipment standards for NBN. We believe a mandatory requirement will be for a 'Simulated-PSTN' connection on customer terminal equipment allowing the continued operation of existing analogue telephone equipment and alarms, during a long transitional period.

4. All IP Connectivity

It makes sense that the PRS industry should eventually move to IP connectivity as an alternative to POTS - the question is, when. Although IP is currently more expensive to implement than POTS, it does offer significant on-going cost savings on calls, and new innovative services developed around IP-based PRS may ultimately lead to increased revenue streams for service providers.

The major obstacles to early adoption of a full-IP solution for PRS are network quality of service (QoS) and network reliability.

> "The PSTN achieves five-nines [99.999%] reliability, equivalent to fewer than five minutes per year downtime, and it handles millions of simultaneous calls. A VoIP network needs to achieve similar levels of reliability and scalability". (MULTISERVICE SWITCHING FORUM TECHNICAL REPORT, MSF-TR-ARCH-001-FINAL, "*Next-Generation VoIP Network Architecture*", March 2003).

Until the time comes when IP connections (including all ancillary modems, routers etc) are as reliable as the current PSTN, (and suitably battery backed-up as required for PRS systems in Australian Standard AS4607), a redundant communications path over the PSTN, or at least the use of multi-path IP, will be essential for mission-critical applications.

An IP connection would really only be an advantage if large amounts of data need to be transmitted (such as video or bio-medical data), if the alarm needs to be continuously monitored, or if call-cost is an important consideration. In a cost sensitive medical alarm market the IP connection appears expensive and unnecessary if a POTS or a reliable 'Simulated PSTN' connection exists.

Large residential aged care facilities and mixed age estates are adopting all-IP networks. All telecommunications and entertainment services are supplied over an internal fibre or cable distribution network and the facility can on-sell telecommunications and entertainment services to its residents. Depending on their design, these private IP networks can provide sufficiently high QoS for a PRS application.

Eventually IP will be the only connectivity option, but for conventional security and PRS alarms the increased data capacity is not required, and the decrease in reliability and increase in equipment cost is hard to justify. The only real driver seems to be compatibility with future communications technologies.

5. Multi-Path IP

Multi-path IP has been adopted by the security industry in an attempt to overcome the reliability issues associated with IP networks. If each IP path (network) does not share any common network infrastructure, so that an outage in one does not affect the other, reliability approaching that of a POTS network is said to be achievable.

Multi-path IP usually includes a wireless path. However, wireless is a rather expensive option for PRS due to the extra cost of a 3G wireless module, and its associated network fees. Scarcity of available frequency spectrum limits the maximum number of wireless users, and price is used as a tool for limiting spectrum usage. However, since a mode-3 wired POTS line is not required, a multi-path IP solution that includes wireless may be cost effective in some situations.

At this time we do not consider wireless sufficiently reliable for a PRS service, except perhaps when a Fixed Wireless Terminal (FWT) fitted with extended battery backup is professionally installed, or perhaps when two wireless modules are provided on two different networks.

At this time, the most reliable and cost effective IP alternative is to use a combination of IP and POTS, where the first outgoing call attempt is made over IP and the second call attempt is made over POTS. Voice communications could be by ring-back or open-voice over POTS.

However, if a redundant path is required to overcome network reliability issues, and if data capacity is not an issue, then why not just use the single most reliable path, which at this time remains POTS?

6. IP Customer Premises Equipment

Currently, connection to an IP service requires various pieces of ancillary terminal equipment installed in customer's premises. Terminal equipment typically consists of a modem and a router which is connected to a home computer. Systems providing VoIP telephony may also include an ATA providing a simulated PSTN connection to an analogue phone or a VoIP phone.

None of this equipment is normally battery backed-up and none would meet the requirements of AS4607:1999.

Some VoIP modems also include a connection for a POTS exchange line, enabling fall-back to POTS when the IP connection is unavailable. The Belkin F1PI242ENau modem, for example, features fallback to POTs and has two phone ports allowing for two phone services, or a phone service plus a fax service, all over a single internet account. Phone calls are automatically routed through the standard POTS line if the IP connection is lost.

http://catalog.belkin.com/IWCatProductPage.process?Product_Id=460610

7. The Codec

To enable audio (voice, or analogue tones such as DTMF) to be carried over a data network, the audio must be converted to digital data at the sending end, and then converted back to analogue audio at the receiving end.

This conversion process is done by a device called a codec (**code**r-**dec**oder) conforming to one of several International Telecommunication Union (ITU) standards. The coder takes 'samples' of the audio waveform at a high rate and converts these to a digital data stream for sending. Digital data streams received by the decoder are reassembled in order, and then reconstructed into the original audio waveform.

Modern Codec's are increasingly implemented as software applications running on specialized microcomputers called digital signal processors, (DSP's).

The quality of the received audio depends on the speed at which the codec's at each end digitise and reconstruct the original analogue audio. Higher sampling speeds provide higher quality transmission, but generate the highest amount of data and require the greatest network bandwidth (and, naturally, also the greatest network cost).

High quality telecommunications networks use ITU-G.711 codec's that digitise the audio into a 64kb/s data channel. This provides good phone quality audio and can also reproduce complex sounds, analogue modem tones, and FAX and DTMF tones.

In an attempt to save bandwidth and cost, highly voice-optimised codec standards were developed – such as ITU-G.723, ITU-G.729 and others. These codec's reduce the amount of bandwidth required by compressing (reducing sound level range) of the audio and by other techniques which highly optimise the coding-decoding process for voice. Very highly voice-optimised codec's are known as 'vocoders'.

The link between the customer premises and the exchange is called the 'last mile connection'. The amount of bandwidth available in the 'last mile' affects the choice of codec, the digitisation characteristics, and the degree of voice compression used.

Multi-standard codec's have been developed that can be reconfigured on-the-run by sending control commands to the network. These codec's can be switched to G.711 mode when a high quality codec is required for transmitting high quality speech, DTMF or data, and switched back to voice-optimised mode for speech.

The choice of codec standard is critical as it determines what, if anything other than voice, can be sent over a network. To make matters even more complex, when calls are routed between different networks, or between networks owned by different providers, often a conversion process occurs which links between different codec standards. Sometimes a first attempt is made using G.711 and if the network is busy (congested) a second attempt may be made using another lower data rate, higher compression standard.

It is therefore difficult to know the exact transmission characteristics of a particular wide-area telecommunications link.

The proposed NBN will have very high capacity and hopefully will not need to limit bandwidth by using voice compression techniques.

Wireless networks use codec's with high levels of voice compression in order to make the most effective use of the limited radio frequency spectrum, and will continue to do so.

8. DTMF Alarm Data over IP/VoIP Networks

PRS alarm dialler's currently communicate with central receivers using a DTMF data standard developed many years ago for the security alarm industry. DTMF is transmitted as an analogue signal containing a mix of two, non-harmonically related, high and low frequency audio tones. The use of two simultaneous audio tones allows the receivers to distinguish between DTMF data and voice, avoiding false receiver activation on voice, or 'voice hits'. However, DTMF receivers are quite susceptible to distortion, background noise, and variation in level between the high and low group tones.

An IP network sends information in packets containing a few tens of bytes of data. The data packets propagate through the IP network from one end to the other in a random manner and, due to network delays (latency), are not necessarily received in the order they are sent. Depending on the path they take, transmission delay may vary between about 60 milliseconds (ms) and 300 ms, or more. The packets are reassembled at the receiving end in a device called a 'jitter filter', and output as much as possible in the order they are sent. Some packets may be received too late to be reassembled in correct order, and some may be lost altogether.

Although not too noticeable in voice communications, these issues become very important when transmitting time-critical data.

Additionally, in order to reduce the amount of network data, data is only sent when there is information to send. Voice activity is detected by an 'activity detector'. However, as the activity detector takes a finite time to identify a valid signal, the leading edge of an audio signal is often cut off. With voice, this is frequently noticeable as a cut to the first syllable; with DTMF, the effect is to shorten the transmitted DTMF pulse.

Alternatively and depending on the network configuration, DTMF tones may not be sent as a digitized audio signal, but rather as a series of 'out-of-band' network level commands to start and stop the play-out of DTMF digits at the receiving end. Network level commands are sent through the network and the DTMF tones are regenerated from these commands at the receiving-end. The latency (delay) of the network can vary significantly and will delay the initiation of the start and stop commands. The effect of this is to significantly stretch the received DTMF tones at the alarm receiver.

DTMF tones sent as out-of-band commands over the 3G wireless network, as is the case when a 3G Fixed Wireless Terminal is used to replace or substitute a PSTN line, can be very significantly stretched to the point where all original data timing is lost.

It is a commonly held belief that because DTMF is so entrenched in modern telecommunications systems, future networks will need to handle DTMF within acceptable transmission delays. The "Network Working Group" is attempting to standardise the handling of signalling and DTMF tones on IP networks. The recommendations in their memos RFC2833 and RF4733 – *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, are now being adopted by network equipment suppliers.

9. Single Tones, Security Handshake Tones, and FSK Data over VoIP

Continuous single frequency tones, including FSK modem data and FAX tones, require the use of a 'clear channel' codec such as G.711. Voice-optimised codec's do not transmit single tones well, causing severe level and phase distortion on the received audio.

Security and PRS alarms using the Ademco Standards use single frequency 'request-to-send' and 'acknowledgement' tones sent from the alarm receiver to the PRS dialler to initiate and acknowledge alarm data transmission. This technique may be incompatible with some VoIP networks and modem's, however the high-quality Optus HFC cable network appears to be able to carry single frequency tones very successfully.

10. Voice Quality over VoIP

To extend the range of voice communications from the client to the alarm dialer, PRS diallers typically use very sensitive microphones and high background noise levels are common. Highly voice-optimised codec's have difficulty processing non-voice audio sources, including constant background noise, (G.711 codec's will not suffer this problem).

In addition, as previously discussed, data is only sent when a person is actually talking and the switching delay in the VoIP activity detector may cut leading voice syllables, affecting intelligibility adversely. This shortcoming would become more noticeable at low voice levels, around the level of the background noise, and at the threshold of the activity detector.

As PRS clients needing help are often remote from the PRS dialler, and background noise in a client's home is common, voice quality over VoIP is expected to be inferior to POTS in a PRS application.

The Utah VHF Society is a group of amateur radio operators who transmit voice and data over a narrowband digital radio system, known as D-Star. "*Observations about the audio codec used for D-Star*" published by the Society, documents their observations of various types of audio sources transmitted over a very bandwidth-limited digital system.

Although using a more bandwidth-limited system than that likely to be encountered in IP-based telecom networks, their observations suggest that VoIP may not provide voice performance in a PRS application equal to that currently obtained through the PSTN. [See appendix A].

11. Alarm Identification using Calling Number Display

Identification of the calling phone number, known as Caller-ID (CLI) or Calling Number Display (CND), could provide a future-proof way of identifying an alarm call. In Australia, private number CND is not available to organisations other than government-provided emergency services, (e.g. police/ambulance), however PRS alarm diallers could be configured to send the standard network activation code to allow CND on emergency calls from the dialler.

Although the Communications Alliance “Industry Code for Caller Number Display” (CLI/CND) ACIF C522:2007 is intended for service providers, we believe the principles could equally be applied to customer equipment:

- The customer must be informed about CND and the operation of the equipment in relation to CND, and
- The customer must have the ability to override any function which enables CND either permanently or for each call.

12. Conclusion

VCI believes POTS remains the best connectivity option for cost sensitive low data requirement PRS applications in the medium term.

IP networks are improving rapidly and their ability to send DTMF and single frequency tones is now very much better than it was just a few years ago. High quality HFC cable networks (Optus), providing a ‘simulated-PSTN’ connection into the modem, have been shown to successfully operate with conventional PRS alarms, although there is an ongoing issue with installation quality and battery back-up capacity.

Multipath-IP, with fallback to POTS, and using POTS for the voice contact, could be a first-step to all-IP connectivity using existing ADSL/Cable networks, though the extra cost and complexity is hard to justify for the PRS application.

Multipath-IP using wireless, as commonly used by the security industry, is not practical for most cost-sensitive PRS applications due to the substantial wireless module costs and ongoing network access fees.

IP/VoIP connectivity alone may be suitable in high QoS all-IP private networks found in some residential aged care facilities and mixed-age housing estates.

Voice contact with PRS clients overt the alarm dialler, especially in the presence of high background noise levels, will probably be inferior using VoIP.

The ability of IP networks to carry machine-generated DTMF data depends largely on the network QoS, the terminal equipment capacity, the choice of CODEC, and if DTMF data is sent in-band along with voice, or out-of-band as network level commands.

The 2009 Federal Government decision to build a universal Fibre-to-the-Home network and the possible separation of Telstra makes it more likely there will be commonality of standards, access technologies and hardware, and higher overall quality of service. However, the technical requirements for the NBN have not been determined at this time.

The widespread view is that, at least in the medium term, NGN's will need to be compatible with existing analogue equipment and will need to provide a simulated PSTN connection which will support existing analogue telephones, and security and PRS alarms.

Our view is that ACMA is very likely to require a long transitional period for the NBN where existing POTS analogue customer equipment is supported.

However, in the absence of any solid information about the technical characteristics of future networks, or the technical requirements for customer equipment connected to them, VCI recommends changing the timings and the format of the DTMF protocol used for PRS, and making other changes, similar to those specified in the new British Standard for Social Alarms. This new protocol extends the DTMF tone duration and has been specifically designed to operate with the packet switched digital telephone exchanges being installed by British Telecom (BT) as part of its 21CN upgrade program.

The suggested modifications are:

- DTMF tones should be transmitted for 100ms and separated by a quiet period of 100ms.
- Alarm receivers will need to identify the format used and reconfigure receiving algorithms to suit.
- Alarm receivers should ignore tone dropouts less than 40ms duration.
- Alarm receivers should ignore artifact tones, (short tones falsely generated within the network or the terminal equipment), less than 40ms in length.
- Alarm diallers should ignore tone drop-outs less than 80ms in Ademco handshake and acknowledgement tones.
- In order to switch the end-to-end network Codec's to G.711, alarm receivers could transmit a burst of 2100Hz tone after call answer and prior to transmitting the first handshake tone. The duration of this

tone would need to be long enough to switch all networks Codec's sequentially, say 500ms.

The same alarm receiver could be used for both existing (short) and new (long) tone formats. The 100ms-on/100ms-off tone periods specified will increase call handling time by around 30%.

Further, VCI recommends that Calling Number Display should be explored, as a future-proof method of identifying a PRS client.

APPENDIX A

A discussion of quality issues when using a highly voice optimized Codec.

"In the case of voice, data reduction is accomplished by preserving only those fundamental characteristics of the human voice that are required to adequately reproduce it. As one might expect, with lower bit rates, this representation becomes less-precise and, inevitably, those deviations from the original source become increasingly obvious.

"Limitations become apparent when these codec's are attempting to encode and replicate increasingly complex sounds, specifically, when other sounds are in competition with the human speaker. As the original voice sounds become "diluted" with extraneous noise, codec's can fail, unable to make sense out of what is being input to them.

"While, in an analogue system, two audio sources simply "mix", the codec will simply "capture" whichever audio source has the most energy. This is readily apparent in close analysis of the "Male-Female voice mix" clip [see website] in which one hears **only** the male **<u>or</u>** female voice at any given instant – never both at the same time. With the highly-intermittent and redundant nature of human speech, it is possible that there will be "holes" in which the sounds of the other speaker can be placed, providing enough information to be able to make some sense out of what is being said.

"In cases where the codec cannot distinguish between the two sources of sound (the speaker, and another voice or background noise) the codec is also likely to produce unexpected results. In the examples above, (see website audio files) one can hear many instances where sounds are produced that resemble neither audio source. In these cases, the codec has mistaken the combined sound as simply noise, or as a random mix of spectral components, and produced its "best guess." Again, depending on the skill of the listener, these can be distracting, or different enough in their sound that they can be readily ignored.

“When presented with a “constant” background noise (such as a generator or siren) there is less opportunity for the speaker's voice to find a ‘hole’ in which a few syllables can be passed by the codec. In this case, it may be that only voice peaks are able to override the offending noise. Whether or not this will yield sufficient information for the listener to be able to understand the speaker depends on not only the extent to which enough intelligible syllables get through, but also the skill of the listener.

“One interesting property with the codec is that, in many cases, the background noise becomes unrecognizable. The implication of this is that the listener may not be able to as easily diagnose intelligibility problems, especially if the listener is inexperienced and is unaware of the problem with the ambient noise.”

“*Observations about the audio codec used for D-Star*” published by the Utah VHF Society.