The IP PABX: System Scarcity Slows Support

Voice over IP as a backbone technology is proving slow to take off, and as Stephen Coates finds, the takeup of VoIP in the enterprise may also be sluggish given the paucity of PABX solutions available.

"Walk into any communications shop, and the resident Gullah will be talking about voice over IP", as Paul Keating might have said had he chosen a different career path. Everyone is talking about it, conferences are devoted to it and every vendor with a product that supports it will quote one research company or another who forecasts that any business that does not adopt it faces imminent bankruptcy.

Yes, voice over IP is all the rage. But what exactly is voice over IP, or VoIP? (It's an acronym that hardly needs decompression.) There are, in fact, four basic applications of VoIP:

- Carriage of voice through an organisation's private IP network in addition to data for which it was most likely to have been constructed;
- Carriage of voice through an ISP-managed, Internet-based virtual private network (VPN);
- Carriage of voice through the Internet; and
- Carriage of voice between terminals of a LAN-based telephone system within the organisation's site.

The first three of these applications concerns long-distance carriage (LAN-based telephone systems are self-contained telephone systems which comprise a highly specialised application of VoIP; see "The Many Applications of Voice over IP, Part 2", *Australian Communications*, April 1999; they will not be discussed further in this piece), which differ in how they use the long distance carriage service.

Organisations adopt the first three VoIP implementations to save on long distance call costs. Whether the organisation has a significant volume of domestic long distance and/or international voice traffic and a WAN using IP, or is currently using the public network, a private TDM or a VPN, it is likely to reduce its call costs by implementing VoIP. And if the network comprises more than two sites, which most do, a WAN which carries packetised voice where the destination address is transmitted in the packet has the advantage that these packets will arrive at the destination PABX without having transited any others along the way. This saves on circuit costs and maintains quality by not decompressing and recompressing the voice traffic along the way.

Long Distance Savings

There are provisos though, two being that network capacity can be increased and the cost of implementation is not prohibitive (unless the network has been overdimensioned).

Long distance carriage corporate VoIP applications require an IP network, typically two or more LANs interconnected by a WAN, to be interfaced to the telephone systems at each site. The WAN component can be either a private network using a TDM, frame relay or ATM service, or a VPN managed by an ISP.

The fundamental issue with any network technology where voice is packetised is quality of service – which means that all packets arrive, they arrive in the correct sequence and they arrive with minimal delay.

With frame relay networks, quality of service is maintained by prioritising voice packets, fragmenting large data packets to not overly delay voice packets, not setting the discard eligibility on packets carrying voice and ensuring the volume of voice traffic is less than the committed information rate (CIR). With ATM, selecting the appropriate ITU-T class (A) and adaptation layer standard (AAL1), and ATM forum service category (constant bit rate), achieves this result.

Figure 1: PABX Connected to Router with Voice Circuits

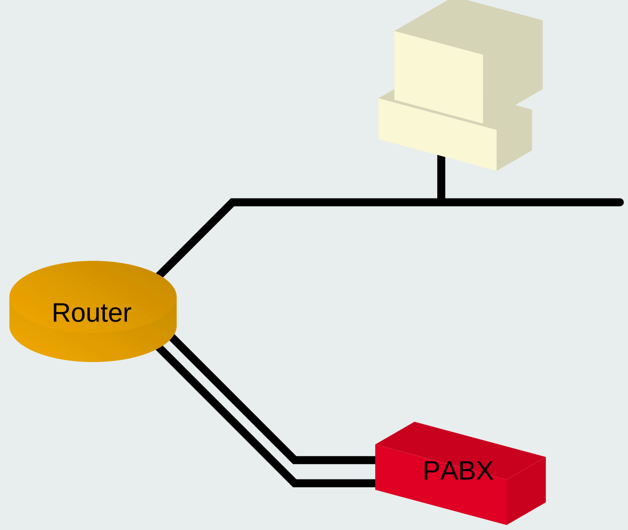
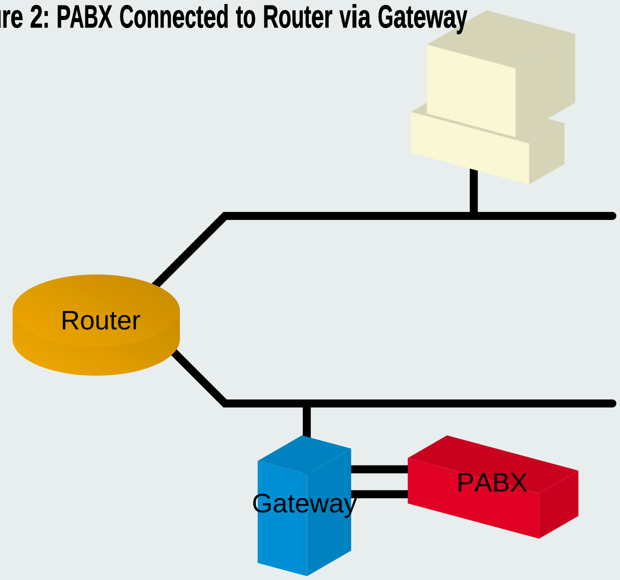


Figure 2: PABX Connected to Router via Gateway



Voice traffic is not delayed with a private fixed-bandwidth network using TDM, so none of these techniques is required. To implement VoIP, the TDM multiplexers would be removed from the network, and routers connected directly to the carrier's network terminating units (NTUs). The routers would have to prioritise the voice packets and, ideally, fragment large data packets to minimise delay.

Turning to the organisation's premises, there are three means by which voice traffic can be connected to the router, frame relay access device (FRAD) or ATM mux, using dedicated voice circuits and a LAN. The first is to use dedicated analogue or E1 voice circuits, as is illustrated in Figure 1.

Routers and Gateways

Clearly, this configuration requires the router to support a voice interface. Many routers do not support voice interfaces and even for those that do, the cost of adding this capability may exceed the price of a new car.

An alternative configuration is to use a gateway from such companies as KBD, Micom and Oki, a type of product which has been available for several years. Such a configuration is illustrated in Figure 2.

Such a configuration is potentially much more cost effective than the first, and only requires the router to support two IP ports and to be able to prioritise traffic from one port over the other.

However, both of these configurations have some significant shortcomings when compared to traditional TDM network capabilities that can be taken for granted.

Consider the situation of an idle circuit between the originating PABX and the local gateway, but no spare circuits to the destination PABX. When the local gateway circuit is idle, it accepts the call and then receives the digits from the PABX. Only when the call is routed through the network to the destination will the busy condition be detected. Because the local gateway has accepted the call, there is no means for the originating PABX to alternate route the call through the PSTN. The caller hears an engaged tone, unaware that the destination extension may not be engaged. Some form of look-ahead routing between the PABX and gateway is necessary, the prospects for the development of which are not very good.

It should be noted that this situation is not unique to routing voice through IP networks – it, too, can occur with a frame relay, ATM or even TDM network where the network and not the PABX performs the call routing.

The second disadvantage is that a trunk or channel on one PABX is no longer associated with a specific destination PABX, let alone a specific trunk or channel on that destination PABX. The problem here is that inter-PABX protocols such as DPNSS and QSIG are designed to be implemented as common signalling channel protocols for which each port on one PABX is dedicated to one other port on one other PABX. These protocols cannot be deployed in either of the configurations illustrated in Figures 1 and 2 on page 86, making extension-to-extension feature transparency next to impossible.

The Need is Not Lost

The need to offer an IP interface has, however, not been lost on the PABX suppliers with about half of PABXs available in Australia and New Zealand supporting such an interface, or planning to do so in the near future. This configuration is illustrated in Figure 3 on page 88.

As listed in Table 1 on page 89, PABXs from Alcatel, ECI (formerly Tadiran), Lucent, Nortel and Samsung offer or will soon offer integral VoIP capabilities using IP circuit cards within the PABXs themselves. The EIC from Interactive Intelligence will also offer this facility, using an integral gateway from Tunda; and the Philips Sopho PABX will also offer VoIP using a Philips gateway that, although external to the PABX, is integrated with it.

Intecom will be announcing a VoIP capability in the near future but it has stated in a position and strategy paper that three objectives must be met:

- Functional hurdles such as quality of service, power, reliability,

Table 1 - PABX Support for Voice over IP

PABX	Software version required for VoIP	Circuits per card	H.323 compliant	Alternate routing	Buffering	Compression algorithms supported	QOS testing	IP address per extension	Networking to other PABXs etc.	QSIG support
4400 from Alcatel, (02) 9951 1222	3.2, due September 2000	30	Yes	Yes	Yes	G.711, G.729 and G.723.1	Yes	Yes	4400 only	Yes
Interactive Intelligence EIC from Call Time, 1300 134 252	2.0, due 4Q 2000	30	Yes	Yes	By gateway	G.711 (µ-law only	No	Yes	Yes	No
ECI Coral from Fujitsu, (02) 9776 4555	10, due 1Q 2001	24	Yes	Yes	Yes	G.711, G.729	Yes	Unknown	Yes	Yes
Samsung from LSP Communications (03) 9872 2926	Unknown, due 01/07/2000	16	Yes	No	Yes	G.723.1, G.729A	No	No	Probably	No
Definity from Lucent, (02) 9352 9000	R7.2	N/A	Yes	Yes	Yes	G.711, G.729 or G.729A	Yes	Yes	Doubtful	Yes
Philips Sopho iS3000 from New Technologies, +64 9 356 1450	810	N/A	Yes	Yes, to the PSTN	Unknown	G.711, G.723.1, G729	Unknown	Yes	Yes	Yes, also DPNSS
Meridian 1 from Nortel, (02) 9857 9333	R25 with MAT 6.6	N/A	Yes	Yes	Yes	G.711, G.723, G.729, G.729A G.729B	Yes	Yes	Yes	Yes

etcetera must be overcome;

- Vendor Interoperability must become reality; and
- The choice of packet switched or circuit switched must be transparent to users.

Apart from supporting an IP interface, these cards are not that different from any other interface card and can typically support as many simultaneous calls as there are timeslots assigned to the card slot. Of course, if more capacity is required, more cards can be installed.

IP addressing is another matter. Some vendors stated they assigned one IP address for the switch, with extensions presumably addressed within the IP protocol. Others stated they assigned one IP address per extension. Emerging technology is never simple.

The routing of a call to a VoIP route within the PABX is no different from a PSTN or private network route. The VoIP interface is either integral to or integrated with the PABX, but compared to a third-party gateway as in Figure 2, the call is still supervised by the PABX as IP packets to establish the calls are transmitted so that if there are no available circuits on the destination PABX, the PABX is able to apply any configured alternate routing. This overcomes one of the primary limitations of IP gateways.

Figure 3: PABX Connected to Router with an IP Circuit

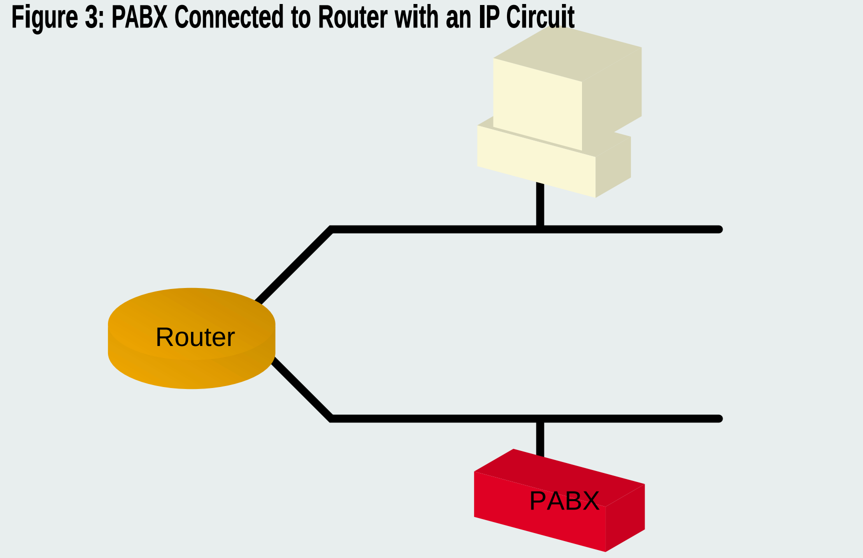
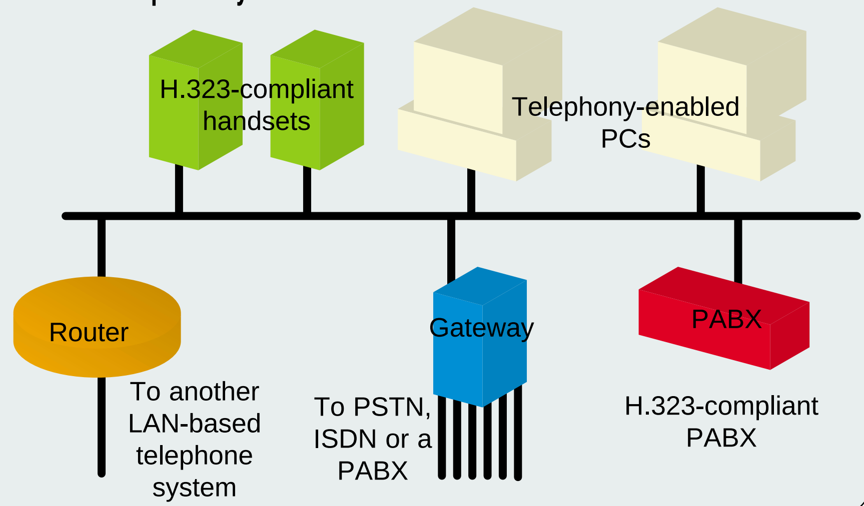


Figure 4: Interfacing an H.323-compliant PABX with a LAN-based Telephone System



Different systems interfaced to, and transmitting voice to each other over an IP network, must all comply with a common standard, and the most widely used standard set for this application is H.323 (see "The Many Applications of Voice over IP, Part 1" *Australian Communications*, March 1999). H.323 is a session layer protocol which defines a set of call control, channel set-up and codec specifications for transmitting real-time voice, video and data over networks, such as IP networks, that don't offer guaranteed service or quality of service.

If two or more telephone systems are compliant with H.323 (version 2 is the latest ratified version), they should be able to be networked using an IP network as is illustrated in Figure 4. All of the PABXs listed in Table 1 claim H.323 compliance.

H.323 Options

However, as H.323 allows a number of optional capabilities (each of which will work only if supported by both endpoints and none of which is required for a supplier to claim that their product is H.323 complaint) a call between two endpoints that are H.323 compliant delivers only the functionality of a standard telephone call.

This is where QSIG comes in. Just like PABX networks using TDM circuits, QSIG can provide extension feature transparency and other capabilities between PABXs networked using an IP network. The 4400, Coral, Definity, Sopho and Meridian 1 all support QSIG, but none reported this being used for networks of dissimilar PABXs. Fujitsu appears to have been the most active in this area, though, reporting networking via Cisco routers, Motorola Vanguard FRADs and Mariposa ATM muxes.

By integrating the gateway function into the PABX, a number of shortcomings of external, third-party gateways are overcome. But there is still the issue of quality of service which is common to all applications in which voice is carried on a packet-switched network. Overall delay must be kept below a threshold of, perhaps, 250ms. This is completely beyond the control of the PABX. The 4400, Coral, Definity and Meridian 1 can send ping signals to determine the delay.

Managing Jitter

What can be managed, though, is jitter, or inconsistent delay. Most PABXs and gateways have, or should have, a jitter buffer to delay early-arriving packets to maintain consistent delay.

This leads to the issue of management. A traditional PABX network using leased circuits is very tolerant of mismanagement. If there are too many or too few circuits, or if some have failed and not been rectified, expenditure will be wasted and there may be occurrences of congestion, but calls will get through and the quality will be good. But an IP network is less tolerant of mismanagement.

If the parameters are not set correctly, or there is insufficient bandwidth, call quality will suffer, along with the communications manager's opportunities for promotion.

For all of the hype surrounding VoIP, the question that must be asked is who is using it. Having extensively researched this question for the article "Voice over IP – Hype or Reality" (*CommsWorld*, April 2000), Shara Evans found that for all the interest, there were very few real world production applications. Intecom's paper cites research from the GartnerGroup and Philips Info-Tech which states that circuit switched technology will continue to play a significant role in the American market for at least five and as long as ten years.

As for VoIP circuits on the PABXs themselves, the others all had only beta trials except Nortel, who cited ten customers using it in Asia Pacific, and Lucent.

VoIP is yet another technology touted to be a revolutionary panacea, that is only enjoying evolutionary uptake. But its use is growing. Watch this space.

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