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TRENDS IN TECHNOLOGY

The Future's Calling

Voice over IP in the real world (and your studios)

By Joe Talbot

It's only reasonable that we begin a discussion of bringing VoIP into the broadcast studio by answering the obvious question: Why? The simple answer: VoIP has become the new standard for telephone service – initially because of cost, and now because of the benefits of open standards and commodity network hardware.

A PBX used to cost from $35,000 to $120,000 for an average suburban radio station. Once you committed to that supplier's switching platform, you were locked into using their phones, service organization and software as well. Frequently, you'd be locked out of your own system. Because of the expense and pain of changing, you'd be stuck with that choice for 5-8 years in most cases.

Of course, some users fooled around with early computer-based VoIP efforts in an effort to save on long distance charges. Back then, the quality was sketchy and you could only call other PC users. You might have to endure pop-up ads for X10 cameras or questionable mortgage brokers. You had to use a headset and your PC would have to be on at all times to receive calls. Over time, the Internet's capability to transport phone calls and streams steadily improved, as did the network infrastructure in most businesses.

Eventually, dominant PBX manufacturers Cisco, Avaya, Nortel and Mitel began making smaller PBXs that used VoIP internally or across LANs, finding that using commodity hardware and the new switching techniques could reduce their own costs and price points. Cell phones became more commonplace, and expectations of audio quality and overall network reliability became quaint and distant memories.

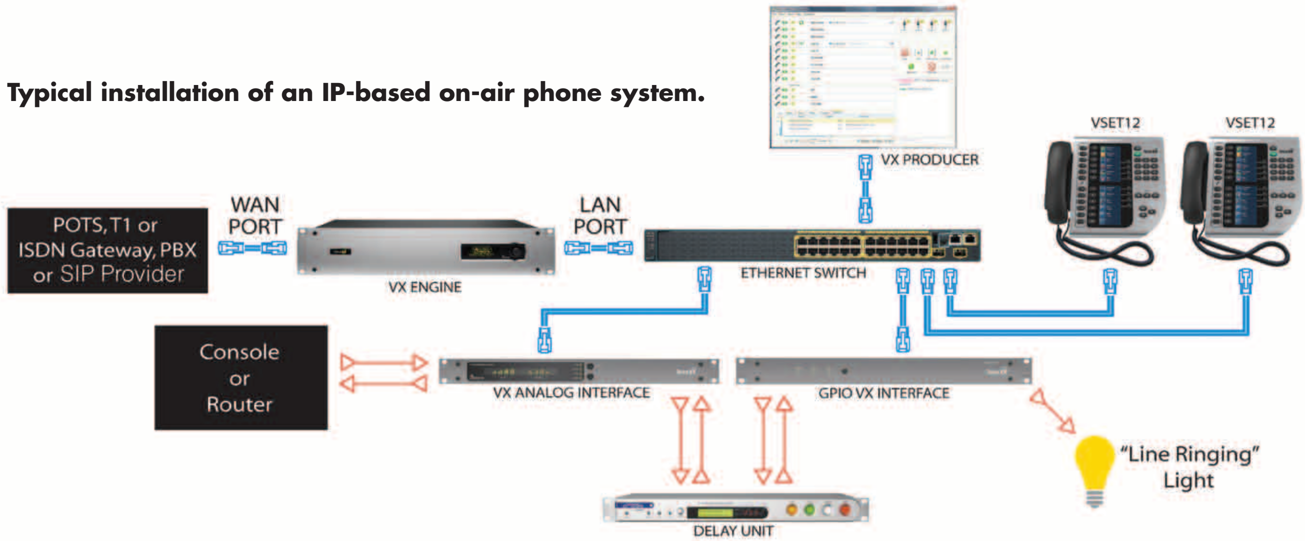
Broadcasters became more IT-savvy, and now had "an IT guy" on call or on site in many cases. That IT guy (and eventually the current crop of broadcast engineers) became more comfortable with fairly sophisticated IT issues such as switches and routers, eventually feeling far more comfortable with network hardware than with the more mysterious legacy telephony hardware.

VoIP advances

As time went on, VoIP was used more and began to perform better and better. While issues remain, it's now beyond simply being merely viable and is well on the way to becoming the standard. In other words, if you ask your phone provider about SIP trunking these days, the likelihood of the response being a blank stare is far less than in the recent past, and far better than ISDN at this point in history.

Even though VoIP switching and transmission are different from that of traditional PBXs, many similarities remain. The audio is still digitized using standard codecs. Users still expect the same features that they always had, such as displays, hold buttons, speakerphones and

Typical installation of an IP-based on-air phone system.



At the 2011 NAB Show, Comrex introduced the STAC VIP, its IP-based phone system.

Telos VSet12

voicemail. The phone still rings at dinnertime to sell you things that you don't want, and you still have to connect to the outside world so that you can call and be called.

Having said that, the differences that VoIP brings the broadcast arena are probably more important than the similarities. With VoIP, you, not the vendor or supplier, can choose the codecs to be used on an individual call or phone type. For broadcast use, higher fidelity codecs like G.722 are easily used, even at the desktop or conference room level. Of course you'll need those higher quality codecs at both ends of the call, but that's easily accomplished if not already the default.

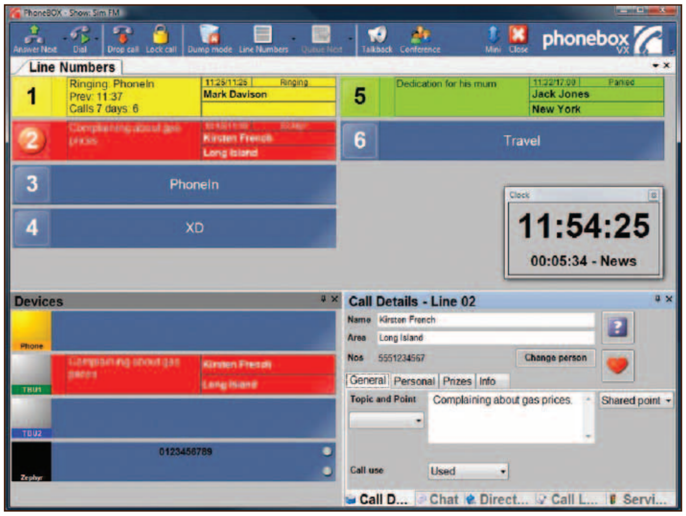
The phone sets, or endpoints in IT terms (because they may not even be physical telephones) use open, published standards to communicate with the PBX and network. This means you can mix and match, buying phones that you like for a particular user or use. Naturally costs of these have declined and the number of choices has risen.

Connecting with the outside world (the Public Switched Telephone Network – PSTN) is now done in many ways. Historically, we've used Plain Old Telephone Service (POTS), an ISDN PRI (a T-1 line set up to use one of the channels as a call set up channel or data channel), or in some cases, even analog E&M trunks for PSTN connectivity.

POTS lines are still the most common and present the most performance challenges. A POTS line uses a two-wire loop that carries DC for off/on hook signaling, 90VAC ringing current, tones for caller ID transmission, call set up, and call progress, and of course speech audio. By its nature, it has loop loss and a fairly complex characteristic impedance, making separation of the transmit and receive audio much more complex and difficult.

Beyond copper

In the current competitive environment, telephone companies are largely ignoring the copper outside the facility that carries POTS lines to your door in favor of selling more exotic and profitable cable-TV-like services. You've probably noticed more scratchy, hummy, lossy lines and, more importantly, noticed the now-lengthy repair commitment times, sometimes stretching weeks-long.



Broadcast Bionics PhoneBOX VX

If we were designing a telephone service for broadcasters from scratch, we would likely specify that it have the following properties:

- A four-wire path; that is, separate transmit and audio paths
- Fast signaling on a separate path
- More control over transmission and routing
- Provisions for carriage of voice, video and any kind of data we could imagine
- It would be dirt cheap VoIP delivers all of these features today; we just have to do things a little differently than in the past to make use of it.

It's completely possible to run VoIP within your own facility and across your own network while still using legacy methods for PSTN connection. You'd be making a mistake by not investigating and taking advantage of some of the new PSTN connection paths and their very aggressive pricing, but you don't have to make that move yet.

I suggest you connect using multiple methods until you're comfortable with a particular strategy. The value of multiple providers and redundant paths in emergency situations also shouldn't be ignored.

In the past, many decisions about phone systems were made because of the phone instruments themselves. People decided on a system because of the look, the feature set or the cost of the phone sets. But now that the system lock in is gone – the requirement to use products from a single vendor – you can choose the end point type based on your facilities' needs and wants without limitations.

End points are telephones, soft phones (computer apps) and appliances. They are what you talk into, and what you hang up. Our company, Telos Systems, has created end points that address the unique requirements of on-air use, including an entire multi-line on-air system (The Telos VX). Others will certainly follow.

The Telos VX emulates VoIP SIP phones, while adding important features that broadcasters need. These needs include phones with advanced screening features, high quality audio inputs and outputs with digital audio processing, a softphone with built-in recorder/editor, multiple program on hold inputs, and routing features that create the ability to move phone lines between studios effortlessly.

Once you've identified what end points you wish to use, a switching platform should be chosen. You need to decide how much control and responsibility you wish to take on or who you give it to. Delegating is good, but so is control. Traditionally, phone vendors performed installation and moves and changes. More than likely, you've taken on some IT duties over the past few years and are pretty familiar with the landscape. I'd say that the question to ask now is, "Do I want more control or less responsibility?"

If you opt for the "less responsibility" approach, hosted IP PBXs are available. You pay monthly, and tell the vendor what you want and how you want it. As in the past, you are constrained by the knowledge, capabilities

Phone Service Guideline

You can save a lot of money by choosing the right combination of services and providers. Here are a few basic guidelines:

- Don't put all your eggs in one basket. Use multiple providers where possible.
- Get rid of all but a few POTS lines. POTS lines are loaded with fees, taxes, and surcharges, and are less reliable than their digitally delivered brethren. Keep a few for emergencies, as the more type of services you have, the less likely any one provider's outage will impact you. Move them to digitally delivered DID-based services over a PRI, IP or a mix of types.
- Keep POTS for fax lines if you're still using fax. The VoIP standard for fax delivery is not widely implemented yet. (In my opinion, it won't be long until fax is gone anyway.)
- Keep ISDN lines if you use those. They're becoming hard to get.
- Port existing numbers or DID number blocks to your provider of choice. This will almost certainly be a competitive provider like a CLEC, cable provider or wireless Internet company.
- Choke lines may not be ported to another company, so evaluate whether you really require them or not. If you're not giving away cars daily, you probably don't need them anyway. (Editor's note: Some phone service providers may still require on-air and contest phones to be choke lines.)
- Don't over-buy services. I recently found a radio station with a maximum of 10 employees on site at any given time that had 48 DID trunks. That's enough for about 48 simultaneous inbound calls and probably 1,000 phone numbers. Insane and expensive.
- Avoid provider contracts more than two years in length, especially with incumbent carriers (Bell companies, etc). Things change, and they need an incentive to perform. Try to get language into the contract that lets you out of the deal if the ownership changes, or if they don't perform as promised.
- Avoid corporate "cram downs" or exclusive contracts. Many individual markets will have opportunities exclusive to their areas. Don't let the fact that your CIO plays golf with one provider's rep determine the fate of the entire company's communications. Seldom are these deals good for anybody but the vendor.
- Pay attention and share information with colleagues. Things are changing fast right now and many factors will affect your costs and capabilities.

and platform of the vendor. The blank stare reaction is a definite possibility here. I've always opted for taking care of my own phone infrastructure; if you're like me, you'll choose to take care of your own switch, which is now easier to do than ever before with free PC-based open source solutions like Asterisk and PBX in a Flash.

Making connections

Once you have end points and a platform, you must determine how you'll connect to the PSTN. Hosted solutions do this for you as the platform and connectivity are bundled together inseparably. There are many methods for connecting to providers, including interface cards for PRI's that plug into your PBX, separate gateway devices that accept POTS, PRI and BRI, and Internet-based SIP providers that will sell you outbound calling services (often unlimited) and inbound DID (Direct Inward Dial) numbers. It allows you to rent a number or block of numbers in most any exchange area of the country or world, and have those numbers mapped to direct phone extensions.

We've used many of these services over the past few years and found them to be very reliable and inexpensive, with phone numbers available for as little as $1 per month. A Google search of VoIP DID providers yields many choices with wildly varying pricing structures and features. We've had very good luck with wholesale providers like Vitelity, Voicepulse, Sipstation and Flowroute, among others. Most will allow you to test with them at no cost to evaluate performance.

If you choose to use this type of provider, you'll need to be sure that you have a solid Internet provider and that you maintain quality of service (QOS) to ensure that any other traffic present on the wire (such as Web surfing or email) won't pre-empt the voice traffic that can't be delayed. Some providers will install a circuit to be set aside exclusively for voice traffic use.

Internet service can be the usual T-1, DSL or cable-based services. Fixed wireless and metropolitan area networks (MANs) are available in many cities, and can be a very good choice.

VoIP has changed the landscape in corporate communications in the last decade. It's easy to install, cost-effective and powerful. Is it right for your broadcast plant, right now? Only you can decide, but the question is no longer if VoIP will come to your studios, but when.

Talbot is a product manager for Telos Systems, Cleveland.

Resource Guide

Broadcast Bionics
+44 1444 473999
www.phonebox.com

Comrex
800-237-1776
www.comrex.com

Telos Systems
216-241-7225
www.telos-systems.com