BRIC Technology

Responding to the Changing Telecom Industry with Reliable, Real-Time, Broadcast Audio Delivery on the Public Internet

I remember when…

I can imagine the conversation: "Back in my day," says the old broadcaster engineer to the new kid, "we didn't have these newfangled ISDN and POTS codecs." It seems like a lifetime ago that every telephone company had its "broadcast division" and equalized loops were easy and cheap to get. When those started going away, analog frequency extenders filled the gap until technology progressed to provide the codecs we take for granted today.

But technology marches on, and not always to the beat of the broadcast industry. Sweeping changes are happening in the telephone network, and they threaten to once again change the way we do radio remote broadcasts.

Migration to VoIP

Anyone who keeps track of technology is familiar these days with Voice-over-IP (VoIP). Companies like Vonage and Skype are increasingly successful in bringing this to consumers, and enterprise level migration to VoIP is gaining momentum. It's popular with users because it can be delivered at low cost and (reasonably) high quality. The efficiencies it provides are not lost on the "old" telephone companies either. Most major local and long-distance telephone providers have announced their intentions to migrate their existing Circuit-Switched-Data (CSD) networks to IP. It will soon become increasingly rare for a phone call to travel entirely over CSD networks. The difference between a VoIP and a CSD- based telephone network is shown in Fig 1. Note that the change is often transparent to voice users, since the analog local phone line remains intact.

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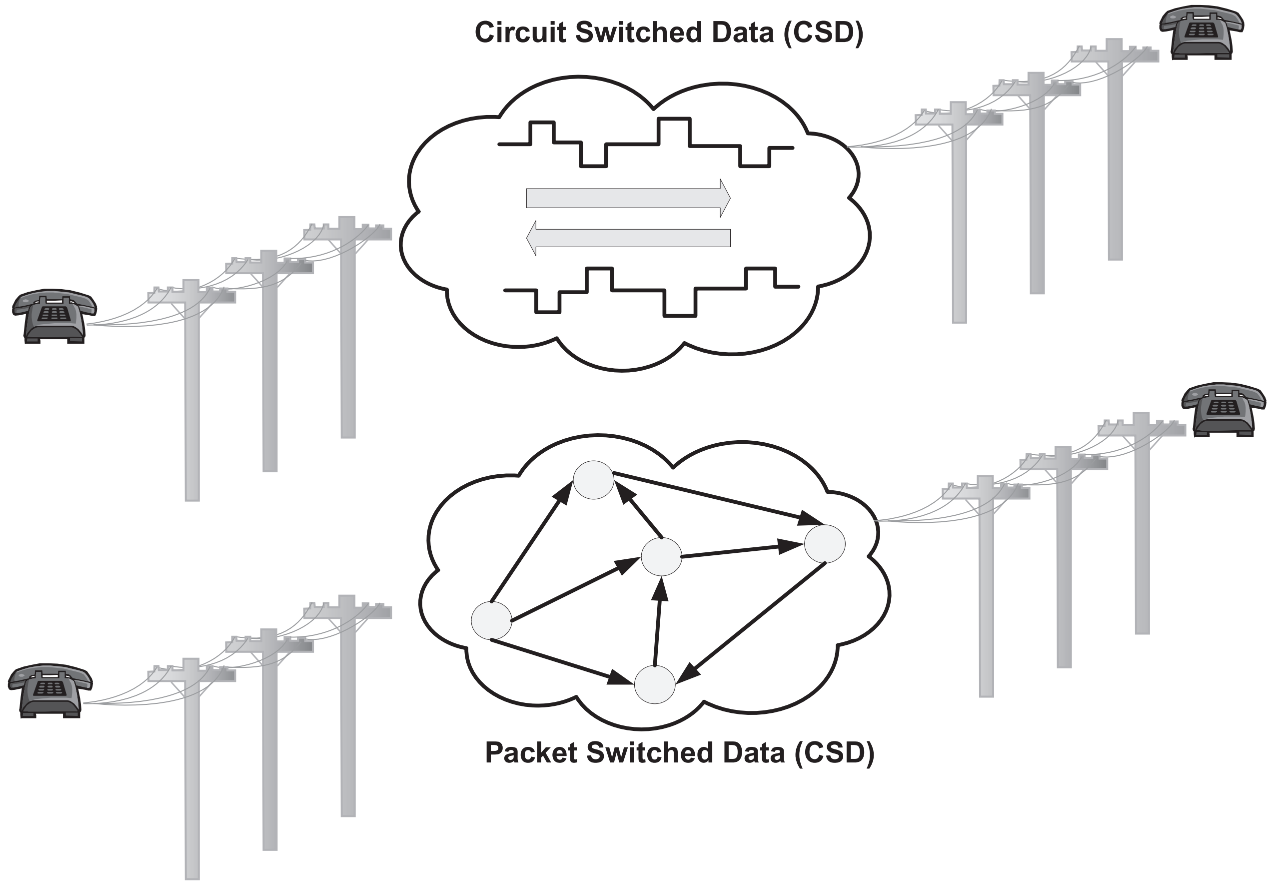

FIGURE 1. CIRCUIT SWITCHED DATA NETWORK VS. PACKET SWITCHED VOIP NETWORK

This can pose a real problem for legacy equipment like modems (like those in POTS codecs) and ISDN gear that rely on CSD, because this gear often performs poorly on networks that utilize VoIP. There have been standards developed to ease this migration by emulating the required protocols over VoIP. But the ISDN network never found its "killer app", and didn't really take off in North America (broadcasters excluded). It's already being phased out in select areas. Likewise, easy access to broadband Internet is making modems much rarer. Pretty much the only legacy CSD technology that still sees heavy, widespread use (especially over long-distance networks) is Fax. And supporting Fax emulation over VoIP links is much simpler and more common than supporting high-speed modems and ISDN. It remains to be seen how much effort will actually be put into supporting ISDN and modem connections over IP networks.

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Even networks that support Modem-over-IP (MoIP) may not support it in a POTS codec-friendly fashion. POTS codecs typically use synchronous modes for their modems, which aren't likely to be as supported as the more common asynchronous modes. Finally, if emulation does work, it's not likely to rival true CSD, since adding a new complex layer to an already somewhat fragile protocol like modems is not likely to enhance reliability.

Simultaneously, access to “IP dial tone” has increased dramatically through widespread deployment of broadband wired Internet (like cable and DSL), 802.11x (Wi-Fi), and high speed cellular data networks. The last year has seen deployment of two new, high speed wireless technologies, Verizon Wireless' 1X-EV-DO network, and Cingular's EDGE network. While on the road, it's much more common now to come across an available Wi-Fi hotspot than a dial-up phone line.

IP CODECS LACKING

In recent years, we've seen the introduction of products coined “IP codecs”. These are typically add-on modes to existing ISDN codecs, utilizing ISDN type of algorithms and wrapping the data into packets for transmission over IP networks. These can be very useful in environments where the network traffic is managed to provide priority to real-time audio. But with a few exceptions, their use on the public Internet has been disappointing. This is primarily due to the choice of audio coding algorithms.

The nature of algorithms like MPEG Layer III and AAC is that the codecs attempt near-transparent audio reproduction, using a fixed compression ratio. This means that their useful data rate is 64kb/s or above for most modes. It also limits the use of data-reduction techniques like voice activity detection and error-hiding techniques like packet loss concealment.

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On the contrary, much work has gone on in the VoIP field to make real-time audio relatively immune to the congestive nature of the Internet. Although the voice coding algorithms usually only supply “telephone grade” audio quality, they tend to degrade much more gracefully under high network jitter and packet loss environments. The hardware used to access these services is not terribly broadcast friendly; they usually emulate a standard telephone or are accessed from a computer desktop.

What is needed is a new way of providing real-time, duplex, high-quality and high-fidelity audio over existing IP networks. As shown in Fig 2, it should combine the superior quality of IP codecs with the stability of VoIP systems. Lastly, it should exist in robust road-ready hardware that can be easily set up and used, even by the non-technical.

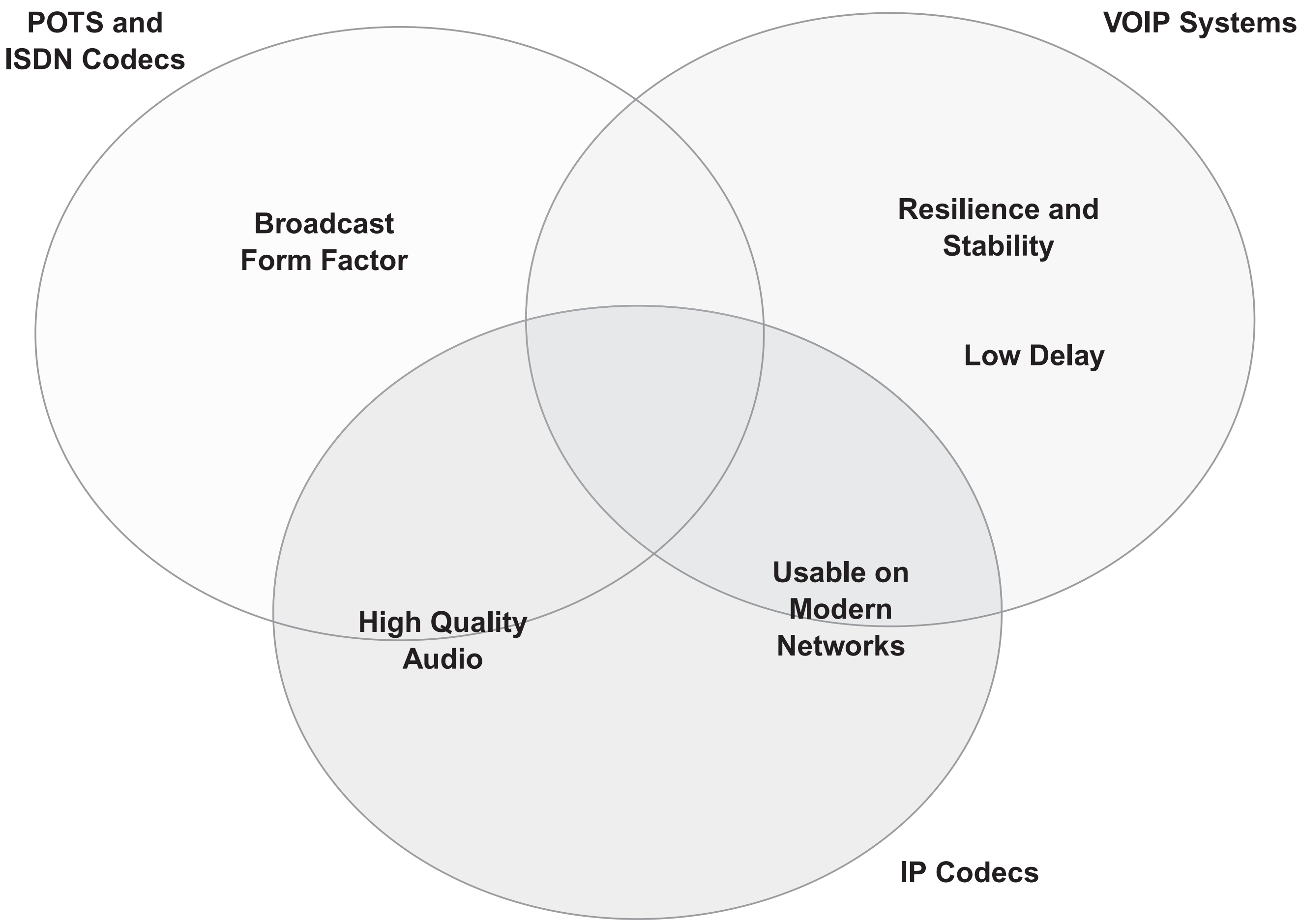

Figure 2 - Characteristics of Existing Technologies Combined in BRIC

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Say Hello to BRIC

As a successor to ISDN and POTS, BRIC (Broadcast Reliable Internet Codec) is the answer to these needs. By borrowing on the strengths of each existing approach, it delivers a way for radio broadcasters to utilize easily available IP networks for real-time delivery of program audio. The first embodiment of BRIC technology is in the Comrex ACCESS series of codecs, which we'll describe shortly. But first, let's introduce the concepts that make BRIC work.

Algorithms

The best way to assure reliable transfer of data over an unreliable network (like the Internet) is to reduce the amount of data sent. Even if your Internet access point is capable of many Mb/s, its path beyond the "last mile" is completely unknown. So BRIC utilizes an audio coding algorithm capable of sending very high quality speech audio (7KHz mono) in a stream under 10 kb/s. That's 1/6 to 1/12 the data rate of an ISDN codec. We refer to this mode as BRIC-UR (for ultra-reliable) and it's the default mode for the ACCESS. Another unique thing about the audio codec is that the data rate is variable. Since IP connections don't have a fixed data rate, it doesn't make sense that your codec should. BRIC-UR dynamically changes its packet sizes based on the complexity of the encoded audio and measured network congestion.

A second choice of audio coding is available in these codecs, called BRIC-HQ (for high quality). This is a remarkable algorithm capable of FM quality stereo at data rates around 24 kb/s. BRIC-HQ can be enabled when the networks used are known to have reasonable data throughput. It should also be noted here that BRIC has a mode that can use point-to-point modem connections (like existing POTS codecs). In this mode, BRIC actually achieves 15KHz stereo over a single dial-up phone call.

One of the biggest strengths of both BRIC algorithms is their low delay. BRIC-UR has an audio coding delay below 100 mS. Although an IP network can sometimes add significant additional latency (which is out of BRIC control), the low delay number on the BRIC codec means even long network delays are often workable. Because audio delay and stability on IP networks are related, simple user controls are available to balance these two parameters.

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Not All Networks are Equal

With its resilient data transfer modes, BRICs are designed to work reliably in very challenging network environments. The codec itself is quite reasonable to listen to on networks where packet loss approaches 30%. It's also important to realize that not all Internet access points are designed to support real-time audio. Many publicly-available networks use Network Address Translation (NAT) and firewalls that limit the ability to connect to VoIP technology to them. There's a third piece in the BRIC family to help with these issues: The BRIC Transversal Server (TS).

BRIC TS

Fig 3 shows BRIC TS. It exists on the public Internet and is supported by Comrex. Use of an external server simplifies the process of getting around NATs and firewalls, since the codecs can maintain a "tunnel" through them to the server, and the server can deliver the current location of any other BRICs that are accessible through the Internet. The BRIC TS can provide information to build a "buddy list" of other BRICs, allowing easy connection — regardless of what type of Internet access they have. BRIC TS can also be very useful in special applications, like when BRIC technology needs to be "bridged" over to legacy POTS and ISDN codecs, or when BRIC audio needs to be archived or distributed to multiple points.

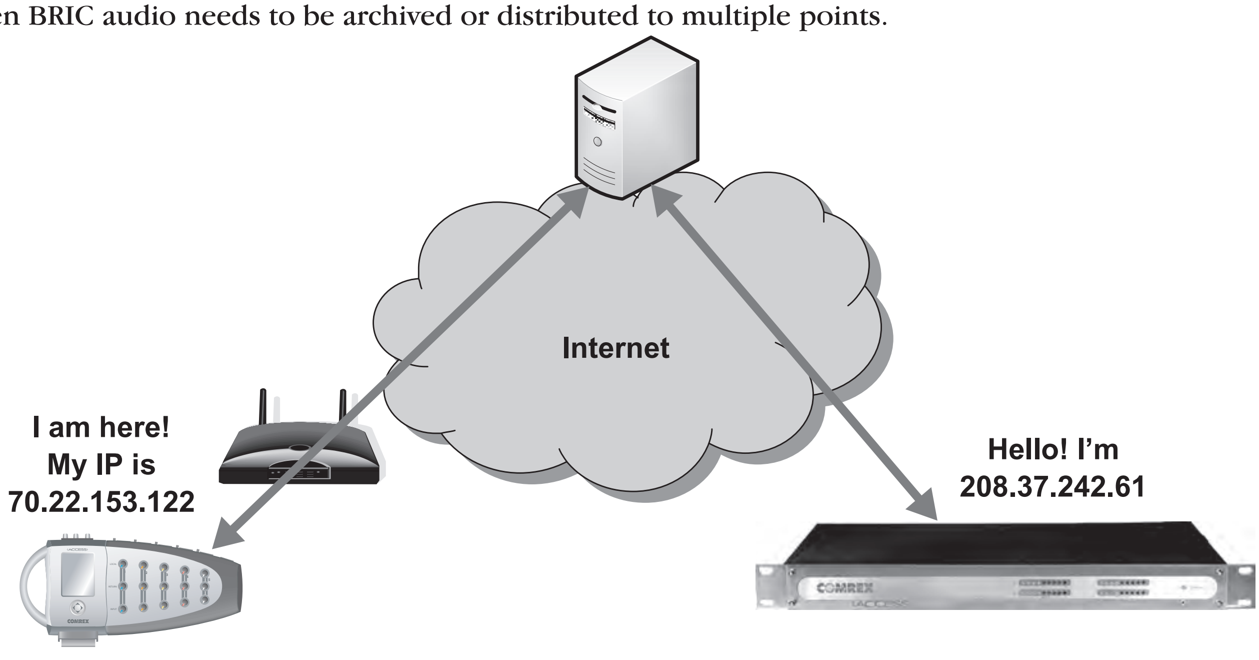

Figure 3 - BRIC Transversal Server

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ACCESS Codecs

The first embodiment of BRIC technology is the new ACCESS codec, introduced at NAB 2005. Like most broadcast codecs, it comes in two flavors, one designed for fixed studio installation and the other for portable use.

ACCESS Rack is designed to be a simple-to-use "always-on" device. Network connections are made via either an Ethernet jack or a modem telephone port (or both). Since it's designed to always be connected to a LAN, user interface is done entirely via pointing a browser to the IP address of ACCESS. An optional module is available for backwards compatibility with all earlier Comrex POTS codecs.

For field use, the ACCESS portable is the really interesting part. This is about the size of a small camcorder, and can run for extended periods on battery power. It includes Ethernet as a default, but also includes a Cardbus (aka PCMCIA or PC Card) slot to allow connection to other networks. These include 802.11x Wi-Fi hotspots, new high-speed cellular nets like UMTS, 1X-EVDO, and EDGE, and dial-up modem connections.

ACCESS portable is designed to be small and hand-held, but can be docked into a multi-channel mixer accessory to provide better stereo mixing and headphone management.

Summary

Migration away from legacy telephone systems is inevitable in the field of live broadcast audio. The BRIC system has been designed to be powerful enough to fill the need yet simple enough to ease the transition. It provides a reliable and rugged way to utilize the public Internet to deliver high-quality audio between sites. Comrex ACCESS codecs, the first BRIC-based products, provides a flexible and rugged platform for use of BRIC technology on a range of networks available today.

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