IP To The Rescue!
(or, What To Do when the Phone Co. says "No!")
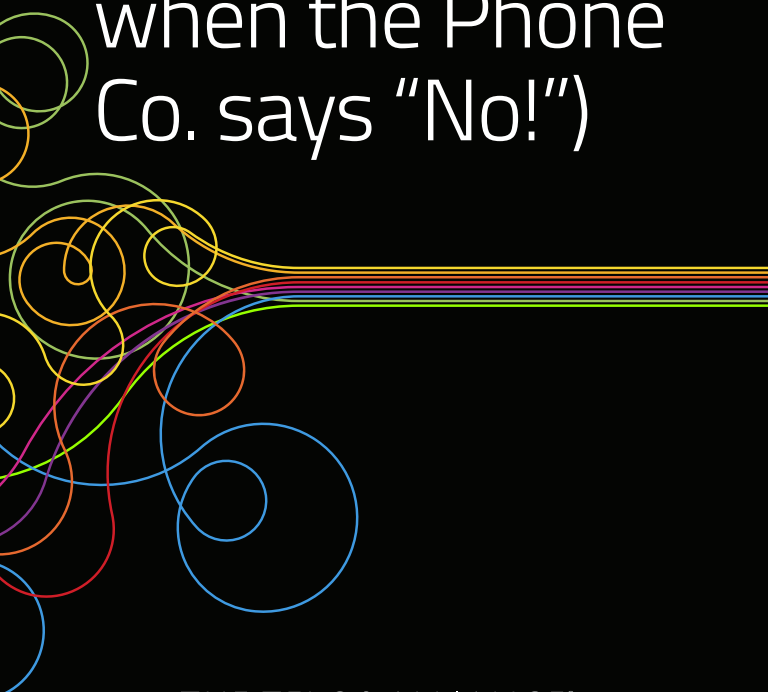
So Long ISDN — Hello IP!
ISDN is nearly gone. But remotes over IP can be even better — here's how.

That was then...

If you're a U.S. broadcaster, chances are you're acquainted with ISDN. Invented at Bell Labs in 1978, ISDN was originally intended as a digital replacement for Alexander Graham Bell's original tech (which was over 100 years old by then.

Although ISDN didn't replace POTS in every home, it gained a huge foothold with broadcasters when Steve Church, a young radio engineer from Cleveland, noticed that the bandwidth of ISDN was perfectly matched to the then-new technology of MPEG-2 Layer 3 audio compression — what would become known as MP3, for short. Steve's invention, the Telos Zephyr ISDN Codec, married the two technologies, and broadcasters bought Zephyrs like thirsty kids at a lemonade stand.

Zephyr was revolutionary because it enabled radio stations to send and receive broadcast-quality audio inexpensively. Remote broadcasts now sounded as if the talent was in the studio, instead of in a tin can; a huge improvement compared to then-standard POTS call-in remotes. Even expensive equalized phone lines couldn't match the quality of ISDN+MP3. Zephyr MPEG remotes via ISDN became the gold standard for radio.

...This is now.

Fast forward 30 years: the telco landscape has changed dramatically since Steve's Zephyr first debuted. Unless you've been living under a rock, you're aware that landline service of any kind is rapidly waning.

Some examples:

- Since at least 2010, US broadcasters have reported increasing difficulty in obtaining ISDN service. Those who still can tell of steeply rising prices for drops. In the UK, British Telecom have announced the end of ISDN (and POTS) in favor of IP services by 2025.[1]

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1 Black, Dominic . "BT to move to an all IP network by 2025", 1 May 2015

- ■ In 2012, AT&T petitioned the FCC to begin hearings for the purpose of determining when the copper PSTN (Public Switched Telephone Network) might be abandoned.[2] This is, of course, the infrastructure on which ISDN services rely.

- ■ In a letter to subscribers dated June 19, 2015, Sprint Communications announced its intention to discontinue all US wireline long-distance services effective September, 2015.
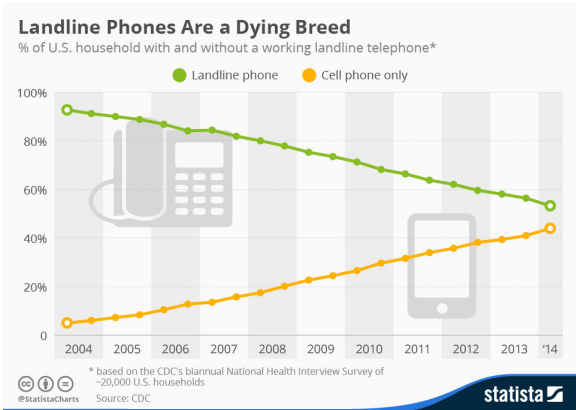
Of course, the public is the one driving this trend, as consumers abandon wired for wireless and IP phone services at an amazing rate. A 2014 study from the Centers for Disease Control shows that as late as 2004, 90% of Americans had wired phone service; 10 years later, the number is at 50% and dropping fast.[3] In fact, the FCC's Tech Advisory Council predicted in 2011 that by 2018, only 6% of the US market would retain wired phone service.[4]

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2 Neuchterlein, Jonathan E., et al. 2012 "Petition To Launch A Proceeding Concerning The TDM-To-IP Transition", Retrieved from http://www.att.com/Common/about_us/files/pdf/fcc_filing.pdf on Aug 4, 2015

3 Blumberg, Stephen J., et al. 2014 "Wireless Substitution: Early Release of Estimates From the National Health Interview Survey, January–June 2014 ." National Health Interview Survey Early Release Program. Retrieved from http://www.cdc.gov/nchs/data/nhis/earlyrelease/wireless201412.pdf on Aug 4, 2015.

4 TECH. ADVISORY COUNCIL, FCC, STATUS OF RECOMMENDATIONS (2011), available at http://transition.fcc.gov/oet/tac/TACJune2011mtgfullpresentation.pdf.

All of which means that ISDN (and POTS) remote broadcasts are going the way of the dodo.

Landline Phones Are a Dying Breed
% of U.S. household with and without a working landline telephone*

● Landline phone   ● Cell phone only

* based on the CDC's biannual National Health Interview Survey of ~20,000 U.S. households
Source: CDC

@StatistaCharts

statista

(Chart reproduced under Creative Commons License CC BY-ND 3.0 courtesy of Statista - http://www.statista.com/chart/2072/landline-phones-in-the-united-states/)

IP Everywhere!

Lest you think that this is all doom and gloom, there's good news: there's a replacement for ISDN, and it's easy to find, reliable, less costly and delivers broadcast-quality audio from nearly anywhere. It is, in fact, the same solution offered by the telephone companies to supplant ISDN and POTS in residential service; sprung from the same technology used by a thriving VoIP industry to obsolete hulking old mainframe PBXs: IP.

Broadband Internet is available nearly everywhere now.

Broadband Internet is everywhere, which makes it attractive for live remotes and STL applications. As of 2013, 70% of American homes had broadband Internet[5], when public spaces and business locations are included, broadband is available to 95% of the US population.[6]

There are plenty of advantages to using IP for remotes:

- The ability to use an existing IP network connection for remote broadcasts, rather than paying for short-term line drops, lowers the costs of high-quality remotes. It also decreases planning and lead time, as IP connections are generally already installed and ready to take advantage of.

- Decreased equipment cost, as codecs for IP remotes are generally less costly than those for use with ISDN. And no "per-minute" connection fees!

- Additional cost savings for long-distance remotes – especially to foreign lands or far-flung states.

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5 "Broadband Technology Fact Sheet", Pew Research Center. Retrieved from http://www.pewinternet.org/fact-sheets/broadband-technology-fact-sheet/ , on Aug 4, 2015.

6 Broadband For America, retrieved from http://www.broadbandforamerica.com/issues on Aug 4, 2014.

■ Less frustration in remote-side setup and deployment of gear, compared to ISDN and its many "flavors" and proprietary provider configurations. (Many an old-timer will recall much time spent on the phone with Telos support or a phone company technician, trying to figure out the correct settings that would make a remote device connect with the studio!)

■ Even though ISDN has been fairly reliable it's stuck at 128 kbps — forever. IP can get you a LOT more bandwidth for higher quality audio. In fact, this is where we're heading eventually (and even now) with linear audio connections of 2 Mbps, with very low delay, available for all kinds of remote broadcasts.

Problems with IP

IP isn't all sunshine and roses, of course. The public Internet has no quality-of-service guarantees, which means that the data packets carrying our remote audio may jitter or be dropped. Bandwidth may vary over the course of a remote. (Some upgraded links can be had with QoS, however - more on this later.)

Jitter, the delay variation in packet arrival, is a part of every Internet connection, ranging from tens to hundreds of milliseconds. In contrast, ISDN has no significant jitter. Sure, you could simply add more buffer to the incoming packets — but this causes significant audio delay, which causes trouble for talent.

Dropped packets are also on the Internet; in fact, it was specifically designed so that overloaded router nodes could drop packets in order to recover (the TCP (Transmission Control Protocol) part of TCP/IP handles this routine). Retransmission recovers lost packets, but detection and retransmission costs precious time.

Finally, the Internet has no guaranteed bandwidth. A high-fidelity codec setting could be negatively impacted if bandwidth declines; a low-bitrate codec can reduce packet loss but sacrifices audio quality.

A broadcast codec intended for IP application must be optimized for that purpose, an integrated system that incorporates technologies designed to counter the shortcomings of the Internet is much more effective than codecs that have not been tuned and optimized specifically for the IP world.

The solution: Z/IP ONE, a codec designed to cope with IP

Fortunately, today's IP codec products are mature; as the originator of ISDN codecs, Telos has been looking ahead and refining products for IP remotes for nearly a decade, culminating in the Z/IP ONE.

The Telos Z/IP ONE.

"Z/IP" stands for Zephyr/IP. It's a broadcast quality, bi-directional codec for IP connections, and equipped with technology designed to combat the vagaries of the Internet — such as:

■   Effective, inaudible packet loss concealment

■   An adaptive receive buffer, with time squeeze/stretch capability to make up for packet lag

- An efficient codec that extracts the best possible audio fidelity from the lowest possible bitrate

- An adaptive codec bitrate that dynamically adjusts to suit network conditions

- A NAT traversal service and online presence server, that simplifies connecting to other codecs by automatically negotiating firewalls and network translation layers.

Here's how these solutions are effectively implemented in the Telos Z/IP ONE IP codec.

First Things First: The Right Codec

Telos' experience in choosing codecs dates back to 1993, when Steve Church partnered with Germany's Fraunhofer Gesellschaft Laboratories to become the first commercial licensee of MPEG-2 Layer 3 – what became known as MP3 – while creating the original Zephyr ISDN codec. Layer 3's optimal bit rate, the rate at which far-end decoded audio became indistinguishable from source audio, was 128 kbps — the exact bandwidth of an ISDN circuit. Our partnership with FhG Labs continues to this day, giving Telos access to a wide variety of the most sophisticated coding algorithms available.[7]

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7 For a fascinating look at the development of the original Zephyr as told by Steve Church himself, see https://www.youtube.com/watch?v=WZqFuvKl2QY .

MPEG AAC has become the modern gold standard for audio coding, and rightly so. Its error concealment capabilities are head-and-shoulders above other codecs; it and the variants used in Z/IP ONE are optimized specifically for use in packet-based transmission paths.

 It works this way: the codec continuously measures and evaluates the spectral signature of the audio to be coded; this is an easy thing to do since the codec also has a time-to-frequency domain transform as part of its perceptual coding routines. When it determines that a packet has been lost, it creates a local replacement using the spectral values of the audio on either side of the lost packet. Volume is adjusted at each end of the synthetic packet so that it matches the "real" audio on either end. It's rather brilliant, and it's very effective in dealing with random packet loss as high as 20%.

MPEG AAC is a very efficient codec to begin with, and the addition of the Spectral Band Replication technique in the HE-AAC (High Efficiency) variant pushes that efficiency even higher. HE-AAC is thus an excellent sounding codec with a small price: a delay of about 150 ms one-way, plus packetizing and buffering overhead. This is certainly fine for one-way remote drops, but you might find it to be a little "slappy" for a two-way live remote.

However, lest you think that a 300 ms round-trip is too much, compare this to a relatively new phenomenon called "buffer bloat", which occurs in many Internet routers.[8] This effect, especially when combined with legacy algorithms such as MP3, can easily result in 300 ms or more of total *one-way* delay. THAT amount becomes awkward, especially for live studio or caller interactions.)

To get that broadcast-quality sound with low delay, we turn to another variant, AAC-LD (Low Delay). It has one-third the delay of HE-AAC (around 50 ms); much better for live two-way conversations. However, AAC-LD is about 30% less efficient than the basic MPEG-AAC codec, which means it's also not as good as concealing packet loss as HE-AAC.

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8 See https://en.wikipedia.org/wiki/Bufferbloat for a description of this phenomenon.

Combining the best features of HE-AAC and AAC-LD yields AAC-ELD (Enhanced Low Delay). AAC-ELD adds the Spectral Bit Replication technology of HE-AAC to the low delay of AAC-LD for a 30% increase in packet efficiency, and truly excellent fidelity. We chose to optimize the AAC-LD coding in Z/IP ONE for 56 kbps and 64 kbps; right in the "sweet spot" for challenging, low-bandwidth connections.

Now, imagine a broadcast codec with a "watchdog" function that monitors the network's maximum bandwidth and automatically adjusts the coding algorithm and bit rate, automatically adapting itself to network bandwidth variance.

Telos has implemented just this sort of mechanism in Z/IP ONE codecs; we call it ACT (Agile Connection Technology). ACT measures end-to-end network performance every half-second, plus packet loss and loss concealment, then adjusts buffer size and the far-end send bit rate to maintain quality. High network capacity results in use of a high bit rate and audio fidelity as high as possible; if available bandwith is reduced, the codec automatically adjusts its bit rate lower to keep audio flowing — and then adjusts it back up as capacity is restored. All of this "on the fly" adjustment occurs without audible glitches.

For example, when "plain" MPEG AAC has been chosen, the Z/IP ONE will automatically switch to AAC-HE if it senses that network bandwidth requires a lower bitrate than AAC will allow. This gives an effective range of 320 kbps down to 16 kbps, as shown in the following chart of Z/IP ONE codec bandwidths:
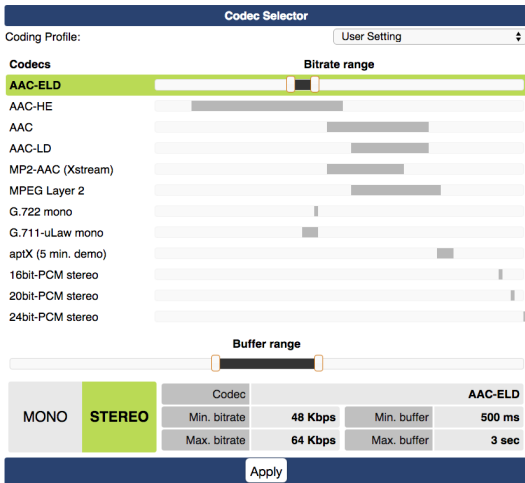
Algorithms and bit rates available in Z/IP ONE

Fine-Tuning The Receive Buffer

As we mentioned already, public IP networks don't offer any sort of guaranteed QoS, so it's impossible to foresee the amount of jitter that will be present. Not only are packets at the mercy of varying network bandwidth conditions, a packet may take a completely different Internet route than the following packet!

To make sure our audio isn't full of glitches and dropouts, a broadcast codec's receive buffer should be able to accommodate the longest delay possible.

But as we've also noted, a long buffer translates to a long delay, so we don't want to just set it for "the longest time available" – we want the buffer to be optimized to the network conditions; moreover, the buffer should adapt to network conditions as they change. This can be accomplished by using the relatively simple method of continually increasing the data flow rate, then leveling off when packet loss is detected. And thanks to Spectral Band Replication technology, those lost packets will go unnoticed in the audio program.

The way Z/IP ONE makes use of this adaptive buffer implementation, the riding algorithm is always trying to reduce buffer length. A fast attack coupled with a slow release allows the buffer to expand quickly if there's packet loss, and contract slowly to seek optimal buffer length; the result is a constant, automatic adjustment for optimal buffer length, with the effect that delay is always the least possible, and audio quality the highest possible.

One more thing: adjusting the buffer length means that audio must be either stretched or squeezed, a function very familiar to broadcasters in audio editing software or profanity delay functions. There is an inaudible stretch/squeeze algorithm built into Z/IP ONE to accomplish just this purpose.

Ready For Transport

There are only three ways to deal with packet loss in most wide-area IP networks:

1. Transmission Control Protocol (TCP)

2. Forward Error Correction (FEC)

3. Concealment

We've already talked about TCP, which works by resending lost packets, but at the cost of much delay to the audio stream and greater bandwidth usage.

FEC deals with packet loss through redundancy, sending both the original packets and duplicates of those originals through the network. If the original packet is lost, the receiver substitutes the copy.[9] While excellent in terms of recovering lost packets, the penalty is delay – *extreme* delay, possibly as great as 600ms depending upon implementation — as well as significantly increased bandwith, as with TCP.

What to do? Effective packet loss concealment is the best choice, coupled with UDP (User Datagram Protocol) and RTP (Real-Time Protocol) for data transmission.

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9 This is a greatly simplified explanation of FEC; for more in-depth reading, try Dr. Rohit Puri's excellent treatise, "Forward Error Correction (FEC) Codes Based Multiple Description Coding For Internet Video Streaming And Multicast".

UDP is a simple, transaction-oriented protocol with very little transmission overhead, which uses datagram sockets to make a direct host-to-host connection. If you've been involved with VoIP installations RTP will be familiar to you; it extends UDP to support media streams by adding a sequence number to UDP packets so that the receiver put packets in order at the far end. RTP/UDP is, in fact, the Internet standard for media transport and gives us what we need to convey low-delay audio streams over IP networks. This, coupled with loss concealment, helps Z/IP ONE to provide very stable IP connections and high quality audio despite varying network bandwidth.

Call Setup

Connecting IP codecs is actually rather simple. Many codecs use Session Initiation Protocol (SIP) for this purpose. If you've used VoIP, you're more than passing acquainted with SIP. SIP is a part of the N/ACIP interoperability standard, an EBU specification intended to allow IP codecs from different manufacturers to talk together.

SIP, in turn, is the carrier for Session Description Protocol (SDP). SDP is the mechanism which tells two connecting units which codecs are available for use, and auto-negotiates the optimum codec choice. Some SIP implementations support technology called STUN/TURN, which helps get through firewalls and NATs (Network Address Translation) layers.

However, being a standard, SIP sometimes lags behind the real world. NAT and firewall implementations have evolved fairly rapidly to meet new security challenges; SIP has moved much more slowly. While the Z/IP ONE has SIP functions (which are required for N/ACIP standards compliance), we found that SIP had problems getting around firewalls and Network Address Translation layers.

To deal with this problem. Telos engineers developed a connection protocol that could be quickly and regularly updated to cope with new security challenges. It is called TSCP (Telos Simple Connection Protocol). It does not rely on SDP, but works with the Telos Z/IP Server, which is a "rendezvous server" that both registers the presence of other Z/IP ONE units and provides a buddy list for groups of related users, helps get around NATs and firewalls, and includes a "relocation service" which allows a Z/IP ONE to be easily found and connected to even if its IP address has changed. Implementing our own NAT traversal techniques allows us the flexibility of updating them when we discover new NAT behavior, instead of having to try to push a standard forward.

Once a Z/IP ONE codec is online, it registers its presence on the worldwide Z/IP Server, generally with a friendly name you've given the unit – "WXYZ Studio ZIP", for instance. Unlike the old ISDN days, there are no SPIDs to fool with, no strings of digits to dial – just look up the Z/IP you want to call using the Connection screen, and you're connected — whether the far end you're connecting to is across town, or across the globe.

You'll have to select a coding algorithm appropriate to your bandwidth, of course, but TSCP and Z/IP Server technology makes it much easier to connect by eliminating the need to manually navigate past firewalls and other security obstacles.

What About Delay?

One good thing about ISDN as a transmission medium is its near-complete lack of delay – any delay to be had came from the encoding/decoding process itself.

In the old days, you may remember that MP3 sounded great, but had about 150 ms of delay, so you generally set up the "send" to the studio with MP3, and configured the backhaul using old-school G.722 (which had a delay of about 20 ms.) After all, it was more important for the talent to be heard, than to hear – right?

Using AAC-ELD, delay drops to around 60ms, which means you can easily deploy it for both sides of the connection. Even if lag in the network adds, say, 75 ms to the conversation, the resulting 195 ms round-trip is about equal to that of modern cell phone connections – eminently workable. And better, more efficient codecs promise to drop encoding times even further in the near future.

Choices, Choices: Picking Networks For IP Remotes

ISDN had a major limitation: its complete reliance on the PSTN. You were at the mercy of your Telco provider for installation, provisioning, service, and also — as a high-value service available from a provider with no regional competition — at the mercy of their billing department.

IP networks, on the other hand, give you choices, and plenty of them.

Nashville, Tennesee's 3rd & Lindsley concert venue sends a live performance to a local radio station via broadband, using the Telos Z/IP ONE codec beneath the console.

The Public Internet

The Internet's biggest advantage is, of course, that its cost is low and it's available nearly everywhere. You don't have to call a month ahead to schedule installation; you just arrive, connect, and you're online. The downside is that the Internet has no QoS , but using the adaptive codec technology built into Z/IP ONE, the Internet becomes a good transmission medium for broadcast remotes. Broadcasters have done successful IP remotes using Internet connections from nearly everywhere, from local coffee shops and shopping centers to amusement parks and even airplanes in flight! And cost remains the same whether your remote location is down the street, or halfway 'round the globe.

Dedicated Links

Even though one of the biggest benefits of using IP is being free of the Telco, there are times when their services might come in handy. In contrast to the public Internet, a dedicated IP link purchased from your Telco via a Service Level Agreement (SLA), which provides a level of QoS to guarantee a set amount of bandwidth, packet loss, jitter, and delay.

Typically an SLA will set out the following spec guarantees:

- **QoS:** delay, jitter, and packet loss limits

- **Network availability.** For broadcast, this should be 99.999%.

- **Scope of service,** *e.g.,* the specific routing of your link.

- **A traffic profile,** defining bandwidth required and any expected burst/peak speeds.

- Monitoring procedures and reporting.

- **Support,** including troubleshooting procedures and response time.

This is an excellent option for STL links, network program feeds and other such point-to-point applications where a higher degree of reliability is needed — but it is not for casual use such as a one-shot remote from a local concert venue or the car dealer across town.

MPLS Service

Another IP service available from your Telco, originally developed for use with low-latency VoIP applications. MPLS stands for "Multi-Protocol Label Switching", a system which attaches labels to incoming packets which describe the network route they will take to reach their destination. Because this method routes packets using short path labels instead of long IP addresses and routing table lookups, the result is significantly decreased transmission delay and, because routers see MPLS packets as a stream, reserving a specified bandwidth is possible. Like the Service Level Agreement described above, MPLS services are obtained from your Telco provider. MPLS provides good "bang for the buck" – it costs more than standard Internet services, but less than a dedicated link or ISDN.

Mobile Data

The increasing speed and worldwide coverage of mobile broadband make it possible to do high-quality IP remotes from the field using software deployed on laptops or smartphones.

3G cellular data networks using CDMA2000 data service with EVDO are good for uplink speeds of 1.8 Mbps; fast enough for a good IP connection to a Z/IP ONE in your studio. 4G networks give us even better throughput, with UMTS/HSPA+ starting at 5.8 Mbps, and LTE delivering speeds of 50 Mbps for "plain" LTE, and up to 1,000 Mbps for LTE-Advanced. (Both EVDO and HSPA have QoS capability in their specifications, but whether this is enabled depends upon the specific carrier.)

Telos' Kirk Harnack demonstrates remote audio via mobile data network, using Luci Live software. Hear for yourself at https://www.youtube.com/watch?v=Wj7evmz2bj4 .

Mobile data networks can be connected to in several different ways:

- Smartphones may be used in conjunction with codec "apps" to connect directly to the studio receiver, either using the handset's built-in mic or a third-party break-out designed to accept pro-level mic input. Telos has partnered with Technica Del Arte so that talent with the Luci LIVE or Luci LIVE Lite app on their phone can take advantage of this capability to connect to their station's Z/IP ONE. Many news organizations have begun using this sort of setup.

- The smartphone may also be "tethered" to a laptop (if this is a part of the subscriber's data plan), allowing pre-recording and editing of field audio, or control of mic/line inputs using a "soft" mixer app, which can then connect to the studio.

- Cellular data USB modems can also be used with laptops to provide connection to the studio.

■ Perhaps most flexible is using a portable "travel router" or "Point of Sale router" to bridge between 3G/4G/Wi-Max wireless networks and an IP codec. Such routers, typically supplied by CradlePoint (and others) offer compatibility with a wide range of USB wireless data modems. Some such routers offer the wireless data modems built-in, requiring only a SIM card from a compatible wireless carrier. A few models offer dual SIMs for more flexibility. These routers typically offer both WiFi and wired connections to one or more devices, connecting them to the Public Internet via the wireless carrier.

WiFi

WiFi is a great way to extend an Internet connection to an outdoor location, where running a LAN cable many yards would be impractical. The most widely-deployed modern WiFi routers use the IEEE 802.11n standard, with data rates starting at 50 Mbps and as high as 600 Mbps — many times more than the bandwidth required for a stable, high-quality IP Codec connection.

We assume you will make good use of this convenience, and a USB-style WiFi receiver is included with each Z/IP ONE unit.

LANs and WANs

You can certainly put an IP codec on a LAN or WAN. But frankly, that's overkill. Modern AoIP such as Axia Livewire+® is a much more flexible and cost-effective solution for inter-plant audio.

Ethernet Radios

An age-old use for codecs has been as an STL link. You could certainly use a Z/IP ONE for this purpose, too. There are plenty of Ethernet radio systems on the market that can be used for STL links, such as those from Dragonwave, Exalt, Mikrotik and Cambrium . Most employ the license-free ISM bands at 2.4, 5.2, and 5.6GHz, and have high-gain antennas; with a direct line of sight, you can use these to make STL shots of many miles (or kilometers). These typically have high bandwidth; a perfect match for a pair of Z/IP ONEs employed in this fashion.

WiMax

WiMAX (IEEE 802.16) was originally developed to deliver fixed wireless service with wireless mobility. (Some folks refer to it as "WiFi on steroids.") Since its introduction in 2005 has been made a part of some cellular carriers' 4G systems, but a more interesting implementation is via standalone gateways which provide Ethernet ports and even RJ-11 jacks for VoIP-to-analog phone service.

Essentially, WiMAX is an Ethernet radio that follows an interoperability standard. They are also capable of making multi-site connections. Bandwidth varies between 17 Mbps and 376 Mbps uplink speed, speedy enough to support any IP Codec connection.

Satellites

Bounce your remote into space and back? Sure, it can be done. It's a rather exotic way to link up, but for those locales that don't have any other way of connecting, an IP-based satellite service and portable dish could be just the thing. Bandwidth is *not* an issue.

Dealing With Troublesome Little NATs

NATs (Network Address Translators) are part of most routers and are used on most Internet connections to allow more than one computer on the inside to share a single outward-facing IP address. (Since random inbound traffic can't get through, NATs also provide a basic firewall function.)

What this means in plain English is that an IP Codec inside a NAT (or a firewall) can't even be *seen*, let alone reached, by a codec in the outside world.

Couldn't we just put our studio's Z/IP ONE right on the Internet? Sure, but you'd be giving up the protection against hackers and random IP probes that are universal in today's connected world. That's just asking for trouble.

The Telos Z/IP Server, a free cloud-based service deployed exclusively for Z/IP ONE users, was designed exactly to counter this problem, while still providing hack-proof connections to the outside for your studio unit. The Z/IP Server provides directory services so that your Z/IP ONE can easily discover other devices. You control whether or not your Z/IP ONE can be "seen" outside your NAT or firewall, and there are visibility levels: public, visible to a specified group of users, or not listed in the directory at all. If you choose to create a group of your own, you can limit access to invited guests only.

The Z/IP Server also features a "buddy list", with a presence server that senses the online state of devices in the list – you never have to wonder whether your remote setup is online or not; you can see it registered in the list.

Another useful service embedded in the Z/IP Server service include NAT traversal capabilities. NATs are usually "symmetric": when a packet stream is sent from inside the firewall toward the outside world, the NAT/firewall opens a corresponding return path, which it keeps open only for a period of time. NAT traversal services keep that connection open to devices that are not usually reachable, allowing incoming calls to get through. For more nailed-up NATs, Z/IP Server can also function as a media relay.

Other useful Z/IP Server functions include geolocation services, and QoS data logging to help keep tabs on packet loss, bit rate averages, and other such link diagnostic information.

N/ACIP – What It Is, And Isn't

Standards are good. Interconnection standards for POTS and ISDN gear allowed those technologies to thrive; why not a basic connection standard for IP-based audio transmission too?

A few years back the European Broadcast Union (EBU) published EBU-TECH 3326[10], in which N/ACIP (Network / Audio Contribution Over IP) was laid out. IP codecs that are N/ACIP compliant, even those from different manufacturers, are able to connect with each other and pass two-way audio.

N/ACIP employs SIP (Session Initiation Protocol) as its framework to negotiating parameters between two devices. SIP also happens to be the basis of VoIP phone gear, so its operation is well-known. SIP is used to set up the call, and then hands over to SDP (Session Description Protocol) for automatic codec negotiation

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10 Read it at https://tech.ebu.ch/docs/tech/tech3326.pdf .

The N/ACIP standard specifies the following codecs:

Required:

- G.711 (the standard telephone codec)
- G.722 at 64kbps
- MPEG-1/2 Layer 2 at 32-384kbps
- PCM linear at 12/16/20/24-bits and 32/48kHz

Recommended:

- MPEG-4 AAC
- MPEG-4 AAC-LD
- MPEG-1/2 Layer 3 at 32-320kbps

Optional:

- MPEG-4 HE-AACv2
- Enhanced APT-X
- Dolby AC-3
- AMR-WB+

This all sounds good so far, yes? However, there are problems with N/ACIP.

For example the N/ACIP standard does nothing to examine the network link between the two connecting units; N/ACIP codec compare their lists of available coding choices during the connection setup and agree on the "best" one – regardless of how much bandwidth that codec requires, and without examining the available link bandwidth. This is not the way you want to start your remote! Neither does N/ACIP look at jitter or packet loss as a setup factor.

While N/ACIP is an important step toward interoperability that all IP codecs should have (and Z/IP ONE does have it), engineers looking at IP codecs should consider that it does not cover all the bases, and many useful features that are engineered into modern codecs – such as NAT traversal, synchronous data transmission, and automatic bit rate adjustment – are not a part of its spec.

Putting It All Together:
4 Steps To A Successful IP Remote

We've covered a lot of ground, but you're probably asking: "When the rubber meets the road, what do I need to do to ensure that my IP remotes are as reliable and sound as great as an ISDN remote?" Great question! The good news is that there are already thousands of IP codecs in use at broadcast plants around the world every day, most of them using public Internet. So, here are what we've found to be the four things you'll need to do in order to create successful, it's-like-I'm-sitting-right-there remote broadcasts via IP.

1: Know Your 'Net.

Modern IP Codecs like Z/IP ONE are pretty great at compensating for less-than-perfect connection quality. From Agile Connection Technology to Network Address Translation to a full complement of efficient, fast coding algorithms, they can make the most of nearly any IP connection. But that doesn't mean you shouldn't do everything you can to give them a little help!

Knowing what your connection will inform the choices you make when it comes time to choose the coding algorithm for your remote broadcast.

- If you're ordering an IP connection (whether it be DSL, MPLS or a fiber drop), get the best, highest-bandwidth, lowest latency you can afford at the outset.

- If you're making use of a pre-installed connection or a third-party "loaner line", make use of online tools to know what it's capable of.

- Use online tools like Speedtest (http://www.speedtest.net/) to help you understand the overall bandwidth of your connection's upload and download links.

- Use an online VoIP tester (such as http://www.voipreview.org/voipspeedtester.aspx) to probe your connection's jitter rate and packet loss. If your connection will support VoIP, it will support an IP remote!

If the results of your testing come back unsatisfactory, you'll know you have some work to do. Read on to find out what that work entails.

2: Optimize Your 'Net.

Most issues that negatively affect IP networks are found in either the configuration of your local router, or in the local use of your network. Here's what you can do to mitigate these problems.

- **Kill the leeches.** Many remote kits include a laptop so that talent has access to the Internet while on site; investigate and make sure that there are no competing connections to the outside. (In other words, no movie downloads, FaceTime chats or BitTorrent downloads during remotes!) These things can suck studio-side bandwidth, too.

- **Set up Port Forwarding in the routers on each end.**
Open router ports that your IP codec uses via your
router's port forward/port mapping features. Even
though the Telos Z/IP Server service can route around
NATs and Firewalls (and even forward the audio if need
be!), making it easier for your codec to connect through
your router generally results in better throughput,
faster connection times and quicker handshake/codec
negotiations. If you're not familiar with how to set up
port forwarding, online resources such as http://www.
portforward.com offer excellent resources — or, call
Telos Alliance 24/7 Tech Support; we're always glad to
help walk you through the procedure.

- **Prioritize audio packet streams.** Another router config-
  uration option present in the majority of today's routers
  allows you to set up stream priorities, so that certain
  types of traffic take precedence over others. Depending
  upon your router's options and capabilities, there can be
  several ways to do this:

    □ Prioritize traffic originating from your
      codec's MAC address.

    □ Prioritize traffic coming through the
      physical router port your codec is attached
      to - set your codec's port to HIGH priority,
      and set everything else to LOW priority.

    □ Prioritize packets by application: since IP
      codec transmission relies upon the RTP
      protocol for its streaming connection, set
      RTP traffic priority to HIGH; your remote
      audio packets will then take precedence
      over all other data traffic.
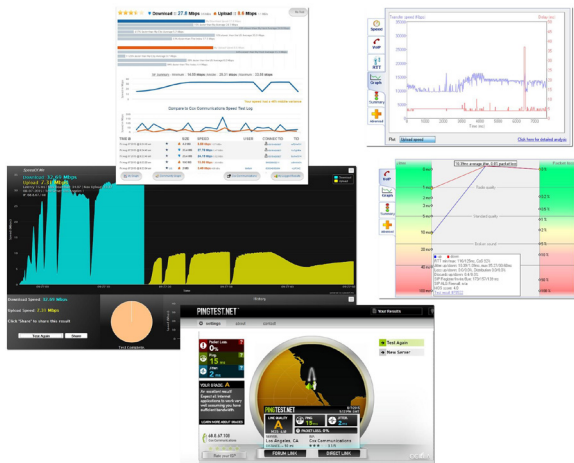
A few of the online tools available for testing IP connection quality.

3: Choose The Right Coding Algorithm.

Knowing your network's capabilities helps you determine which algorithm is best for your remote. Each coding algorithm supplied in your Z/IP ONE is optimized for different combinations of audio quality (bit rate) and stream latency (buffer size). Here's a list of the choices available in your Z/IP ONE, and what sort of application each is best suited for. You determined your bandwidth in Step 1; now choose the highest-quality your connection will support based upon your needs.

- **G.711** – plain old 3 kHz POTS telephone audio quality. For making phone calls between far ends. Consumes a fixed bandwidth of 64 kbps.

- **G.722** – "HD Voice" 7 kHz VoIP telephone audio quality. Better voice calls! Also 64 kbps bandwidth.

- **AAC-ELD** – Choose AAC-ELD for voice remotes on connections with very limited bandwidth. Variable bit rate of 16 – 64 kbps.

- **HE-AAC** – Choose HE-AAC for better quality on higher bandwidth connections. Great for voice and music; variable bit rate of 16 – 96 kbps.

- **AAC** – The "Gold Standard" for medium-to-high bandwidth connections. Choose AAC for superb audio quality with any program material. Best for remotes where high audio quality is desired and backfeed delay is not a concern. Variable bit rate of 96 to 320 kbps; 128 kbps is the "sweet spot" of audio quality and bandwidth consumption.

- **AAC-LD** – Choose AAC-LD for two-way remotes with lots of interaction between studio and talent. Delivers excellent audio quality for voice and music, but with lower audio latency than "standard" AAC. Variable bit rate of 96 to 320 kbps.

- **Enhanced apt-X® (optional)** – An optional, extra-cost codec choice available to Z/IP ONE clients. Suited for very high-bandwidth links where high-quality audio and low latency are required. Fixed bit rate of 512 kbps.

4: Make backup plans.

Redundancy is good. Even if you've got an excellent IP link, don't assume it's failure-proof – have a fallback whenever possible. Modern IP service is very robust, but it never hurts to have a contingency plan. Got WiMax? Think about a DSL drop for back-up. Using 4G to send audio back to the station? See if there's a WiFi connection nearby as well.

Useful Links

Below is a list of analysis tools and resources we've found useful:

- https://youtu.be/Ri3mweYQikg?list=PLrVvkZpBBU38v-S4E1Zh6WpiQdGVVCWb7k - 'Three Keys To Great Audio OVer the Public Internet" video from Kirk Harnack

- http://testmy.net – network upstream/downstream speed tester. Can be set to run every hour, automatically, and average results over a day's time.

- http://speedof.me – an all-HTML5 speed tester.

- http://www.voipreview.org/voipspeedtester.aspx - Excellent resource for IP suitability testing. If Your connection is acceptable for VoIP, it will be acceptable for IP remotes as well.

- http://pingtest.net – a standalone tool for measuring packet loss and jitter.

- http://n1.netalyzr.icsi.berkeley.edu/analysis/ – Very detailed analysis of internal and external network status and port configurations.

- http://www.bufferbloat.net/projects/cerowrt/wiki/ Quick_Test_for_Bufferbloat – a variety of tests to determine whether your network router suffers from "buffer bloat", along with suggestions for solving it.

About Z/IP ONE

With Telos Z/IP ONE, you don't have to compromise audio quality for a solid connection. Z/IP ONE helps you get the best possible quality from public IP networks and mobile phone data services — even from connections behind NATs and firewalls.

Telos collaborated with Fraunhofer (the developers of MP3) to develop a unique operating algorithm that adapts to changing Internet conditions on the fly, helping you maintain quality and stability. It's called ACT (short for Agile Connection Technology), and only Telos has it. Using ACT to sense and adapt to the condition of your IP link, Z/IP ONE delivers superb performance on real-world networks, giving you reliable audio despite varying network conditions — and without the need to fiddle with settings or codec choices. When the bits are flowing smoothly, you'll benefit from the lowest possible delay and the highest possible fidelity. If congestion starts to occur, Z/IP ONE automatically lowers bit rate and increases buffer length to ensure maximum audio quality.

Another way Z/IP ONE extracts excellent quality from not-so-excellent IP connections lies in its use of the Advanced Audio Coding-Enhanced Low Delay (AAC-ELD) codec, which gives excellent fidelity at low bitrates with nearly inaudible loss concealment and very little latency. (You have your choice of standard high-performance codecs too, including HE-AAC-HE, AAC-LD, MPEG4 AAC-LC, MPEG2 AAC-LC, G.711, G.722 and even linear PCM. Enhanced apt-X is optional.)

Z/IP ONE's front panel is friendly and simple to use. A built-in web server provides remote control and easy configuration via Web browser. Our exclusive Z/IP Server service, free to Z/IP owners, lets you easily get around NATs and network firewalls for fast connections to your favorite locations. Around back, there are convenient XLR ins and outs for Analog and AES/EBU audio, a Livewire+® port to work with your Axia AoIP network, LAN and WAN jacks, and a parallel port for RS-232 serial and GPIO contact closures, with functions time-synchronized end-to-end. Z/IP ONE will even make an automatic reconnection to your last location in the event of a power loss or network outage during a remote.

To download a brochure or find a Telos Distributor, visit www.TelosAlliance.com/zip-one.

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Thanks for your time and attention.

Best Regards,
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