

USING T1, ISDN AND DSL TELCO LINES IN CONSOLIDATED RADIO FACILITIES

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Modern radio studios are digital, computer-assisted, and often consolidated. Modern telecom connections are digital and often high capacity, bearing little resemblance to the analog lines of old. Today's studio telephone interface gear must be ready to link these worlds, delivering audio quality and control flexibility. This article outlines the available digital telephone connection scenarios and describes requirements for a system that will capably serve today's multi-studio facilities.

For decades, radio studios were pretty much the same: Analog consoles were connected to audio sources with good ol' Belden 8451. You had your basic FM/AM combo with two control rooms and a couple of production studios. The biggest innovations were slide pots and punch blocks. How the world has changed! Now most audio is likely to come from a computer and there may well be 8 – or 80 – studios in one place. Mixing consoles at new installations are likely to be digital, and will soon be universally so.

The one stick-in-the-mud has been the connection to telephone lines for call-in use. Few question the power of having real people's voices on the air, and dial-up telephone is the way to get this done. There are maybe a billion phones in the world – and all of them can become an audio source with a few button punches. The massive and ubiquitous telephone switching network as an extension of your studio. Nearly every facility has some way to hook into the phone network, almost all using the analog lines invented in A. G. Bell's time. Think about it... blasts of 100 VAC to signal an incoming call? How long has it been since you listened to clanging metal to know you had a call? Yet, that is probably the way your studio gear knows someone wants to talk.

This despite the fact that the telephone network is nearly all digital, with only the "last mile" connection from the Central Office being analog.

DIGITAL VOICE TELEPHONE SERVICES

You may be familiar with the Telco's digital possibilities because of your use of MPEG codecs over ISDN. In that case, the network provides a simple 64

kbit/sec end-to-end digital path. MPEG encoding is powerful, in particular MP3 and the new MPEG AAC, so it's possible to convey full-fidelity stereo using ISDN's two channels. ISDN works all over the world with standard and familiar dialing procedure because the ubiquitous voice network is inherently comprised of 64 kbit digital pipes.

ISDN for Voice

We are so used to using ISDN for codecs that we forget that this is not what it was created for. The original idea was that a user would have a voice call going on one channel while a computer data connection was on the other channel. ISDN was designed primarily to hook into the standard voice network and interwork with the enormous quantity of analog phones out there. In addition to providing data service, ISDN was to offer a way for Telcos to give users a variety of modern call processing capabilities. For example, you can connect up to eight phones on a single line, with each having its own directory number for incoming calls.

For normal voice telephone calls, a coding method much simpler than MPEG is used. Phone engineers invented it in the 60's when processing power was not so available and cheap as it is today. This is called *instantaneous companding* and allows the natural 8-bit depth to represent approximately 13 bits dynamic range. The sampling rate is 8 kHz, with the usual top audio frequency being 3.4 kHz. While not high-fidelity, this *is* digital audio – indeed it was the first application for digital audio technology.

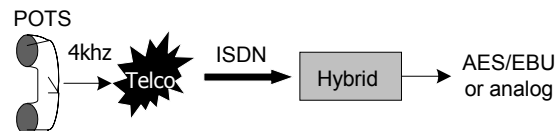


Fig. 1: ISDN interworks with analog POTS lines. The Telco CO provides the required conversion. The studio hybrid separates the send/receive as usual, and provides sample rate conversion and other functions.

ISDN has become widely used for standard voice connections in many European countries, but has failed to catch on in the USA for this purpose. Could be the cost of phone sets, ordering troubles, or the use of confusing SPID numbers was the cause. Nevertheless, ISDN is near universally available and works perfectly well to get a pure digital connection from the telephone network to our digital studio gear.

The advantages are clear and compelling:

- **Superior audio quality** All kinds of interference from power lines and other noise sources plague analog lines; while digital ones are immune. One side of Telco A-to-D and D-to-A conversion is eliminated. Since Telco analog converters are usually not so good, this helps a lot with signal-to-noise and distortion.
- **Near perfect hybrid null** The studio side of the connection is now “4-wire,” with independent send and receive paths. We still need an adaptive hybrid because the caller’s analog connection has mixed send and receive audio and these need to be accurately separated. But a careful digital hybrid design combined with the digital connection on the studio side can reduce leakage to near perfectly absent.
- **Better call setup and control** The main advantage is eliminating the common dial-tone blast problem, where a host punches up a call only to be greeted with rather unfriendly dial tone. This happened because the caller hung-up and the studio phone system didn’t know it. The Telco CO took too long to signal the status change, so there was an ambiguous period when the two ends were not in sync. With ISDN, this never happens because the state of the call is signaled instantly.

T1 and Primary Rate ISDN

So far, we have been talking about ISDN Basic Rate Interface, with its two 64 kbit/s channels. But that is certainly not the end of the digital telephone service story. In the 1960’s the Bell System was running out of copper capacity, probably owing to the effects of the baby boom on population and the rapid growth of the American suburbs. To solve the problem, engineers invented a way to carry 24 phone calls on a wire pair. These were named *T1 lines* signifying the lowest level scheme for digital call transmission. Here we have 24 channels x 8 bits x 8 kHz = 1.536 Mbit/s total capacity. T1s are still widely used in North America, despite their age and the use of primitive signaling methods derived from analog pulse dialing. For example, dialed telephone digits are conveyed by wiggling a bit as if it were being controlled by a rotary dial. “Wink start” and “hook flash E&M” begin and terminate calls. Not exactly computer age, but functional. The Telcos save a lot of copper: only two pairs are needed for all of the

channels, one for each direction. This compares to the 24 pairs that would be required for analog.

In Europe, E1 lines are used. These are identical to T1s, but with 32 channels, rather than 24.

Primary Rate ISDN modernizes T1 and E1 lines. One of the channels is taken over for call-control data, so you have either 23 or 32 channels remaining for voice. This 64 kbit/s data channel allows more sophisticated information to be transmitted, rather than the simple dial pulses.

T1s are widely available and used in the USA, and costs are falling dramatically since the FCC opened up competition in 1996. Larger cities have a dozen or more companies competing to provide them. The usual application is a direct connection to a long distance carrier, bypassing the incumbent Telco and saving the around 2.5 cents/minute that the Telco would collect as its share for routing and connecting the call locally. There may also be another T1 for the local service. The PBX uses *least cost routing* to send the long distance calls directly to the T1 bound for the long distance carrier and the local calls to the T1 going to the local Telco CO. Of course, it is possible to have both local and long distance on a single T1. You save the recurring base expense for one of the T1s, but calls will usually cost more per minute.

Carriers and customers in the USA are slowly migrating from T1 to Primary Rate ISDN. Since the underlying transport is T1, changing means installing a new card in the CO and PBX to support the more advanced signaling. But the wire pair and terminating equipment can often be re-used. ISDN PRI is readily available in most of Europe and is widely and routinely used to connect PBXs to the telephone network. ISDN PRI is better than T1/E1. For example, Calling Party Number (caller ID) is not generally carried on a 1-800 long distance T1 trunk, while this is routinely available on an ISDN PRI.

When you buy your T1 or ISDN from someone other than your incumbent Telco, you are still likely to see their familiar logo on the trucks in your driveway come installation day. That is because they will probably still provide the raw wire pairs and some terminating equipment. The company with whom you signed the contract will provide the switching and billing. But this is not always the case; sometimes a vendor may have its own cable and terminal, this being more probable if you are in a big office building in a major city center. And other arrangements are possible, owing to the variety of companies and services competing for your business. Pricing varies considerably as well, so it pays to shop and compare.

Fiber and SONET

Increasingly, fiber cables are replacing copper for T1/E1 and PRI ISDN delivery. From the user perspective, nothing changes – the Telco will provide the boxes to convert the optical connection to the standard

copper T1-style RJ jacks and all your gear connects as if the delivery were on copper pairs.

If you are a really big user, or are simply in the right place, you may get your feed directly from a tap on the Telco's *SONET* (*Synchronous Optical NETWORK*) Ring. This is fiber writ large: an OC-1 has a 51 Mbit/s data rate, while an OC-3 can handle a whopping 155 Mbit/s. Because SONET uses an add/drop scheme, you can break out (and pay for) only the capacity you need in T1-sized chunks.

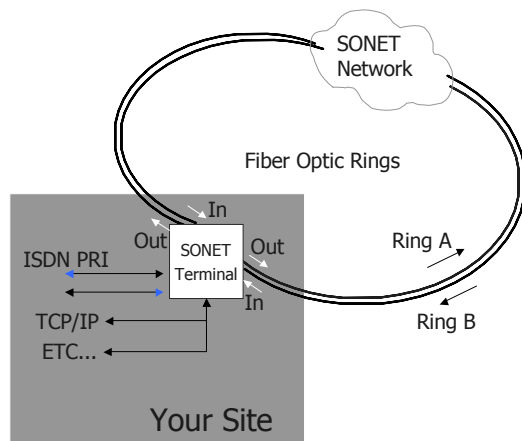


Fig.2: SONET Ring connection may be the ultimate hook-up.

Voice Over IP

There is yet another digital telephone connection possibility on the horizon: *Voice Over IP*, or *VOIP*. All of the systems described so far are so-called *Circuit-Switched* connections over *channelized* bit pipes. That is, the calls are handled as continuous streams of bits at a 64 kbit/s rate. While a call is connected, the channel is nailed-up and dedicated to this one call. The Internet uses a fundamentally different way to convey its bits: *packets*. These are groups of bits organized according to Internet Protocol (IP) that include a header containing the source and destination addresses. Using these addresses, routers, placed at various nodes, direct the packet to the specified end point. Voice signals can be encoded and packetized for transmission over the Internet – bits are bits, after all. IP packets offer flexibility compared to circuit channels, but at a cost.

Current VOIP has a deserved bad reputation for quality. The main problem results from the current state of the public Internet, where there is no guaranteed bandwidth for a call: all packets on a segment contend for capacity, and it is often the case that there is more demand than space – making for choppy audio. But there is no technical reason why bandwidth can't be assigned and assured, and next-generation and private IP networks will almost certainly support this service. The second problem is poor audio quality – even when there is enough bandwidth – caused by the compression applied to conserve bits. Even the core 64 kbit/s in a standard voice

channel is too much for modem connections to handle, and packetization adds a lot of overhead – as much as 100%. So VOIP designers must use aggressive compression to reduce bit rate. Typical is 9 to 16 kbit/s, and this doesn't do much for the audio! But future networks will have plenty of capacity and it should not be necessary to compress to the extreme.

The main appeal of Internet phone calls now is cost, but it is not clear if high-quality VOIP will preserve this advantage. Packets are less efficient from the perspective of bit consumption, owing to the addition of headers, checksums, etc. This can be made up with compression, but to the detriment of audio quality. If providers give quality guarantees comparable to traditional phone service, they will consume more bits per call than the current method, so might need to charge customers at close to circuit-switched rates.

Delay is another issue. Because it is tremendously inefficient to put only a single audio sample in a packet, multiple samples are buffered, accumulated, and packed into each packet. This requires buffering, which causes delay in the audio. Then, more buffering is added to cover network delays.

Nevertheless, IP's flexibility is likely to mean that it will eventually be widely deployed for voice, owing to the advantages it brings:

- Common cables, switches, etc. can support both voice and data.
- IP routers are inherently simpler and will be cheaper than circuit switched CO equipment.
- Routers and associated networks are more readily scalable.
- A wide range of media types and different audio quality levels are easily possible simultaneously on a single network.

The computer industry has tremendous financial clout and R&D resources with which to challenge traditional telecom vendors. Cost per bit for transmission is steadily and dramatically falling, making IP's inefficiency less a problem, and its flexibility more an appeal. Some telecom pundits say "bet on bandwidth," comparing this to the bets on CPU speed increases that created winners in the last computer industry round. Today this is all very interesting to observe, but pretty much academic for professional users. Until the problems are sorted out and serious service levels are available, VOIP is not an option for broadcast applications.

xDSL

When VOIP becomes mainstream, it will probably be delivered over xDSL lines. Like ISDN, xDSL is a way to transmit digital data over the Telco's existing copper wires. Because it is a newer development, it offers higher bitrates – ranging from 512 kbit/s to 6 Mbit/s.

The term *xDSL* specifies a Digital Subscriber Line using any of a number of technologies. ADSL and HDSL are the most common variants, and there are around a dozen incompatible implementations. Incidentally, some people consider ISDN to be an *xDSL* mode.

ADSL uses fully reliable ATM as its underlying protocol, not IP, so it naturally can work to provide ISDN-like voice connections without the problems described above for VOIP. But that is not how it is being deployed today. Most often, on the user terminal box there is an analog voice port and an Ethernet IP jack for the Internet – no digital connection to the voice network. It *could* be possible to use the Internet for VOIP, but this would come with all the usual problems.

So, What About ISDN's Future?

With the rise of ADSL, an interesting question (in the USA) is “What is the future of ISDN?” Some Telco clients have reported to us that their Telco is trying to steer them to ADSL when they call to order ISDN lines, the Telco rep being apparently convinced that the only application for ISDN is Internet hookups, for which ADSL is undoubtedly superior. Until VOIP becomes routine and reliable, it is unlikely that ISDN will go away. That's because it offers the only way to get a reliable and normal audio quality digital connection into the voice phone network. One possible direction is what we see in Europe, where some ADSL terminals have ISDN BRI jacks.

FACILITY-WIDE STUDIO TELEPHONE INFRASTRUCTURE

When radio studios came in pairs for an AM/FM facility, a dozen studio Telco lines was generally enough. But the now-common consolidated plants could use 100. With this number, telecom providers are certainly going to want you to take the service on T1s or ISDN PRI. Which should be just fine for you, as well, because you will get all the advantages of digital connections.

There will also be economic benefits. A channel on a T1 costs less than an analog pair. Per-minute long distance costs can be lower, if you bypass the Telco and go straight to your LD carrier. Because the pool of lines is shared to all studios, you pay only for capacity necessary to support the peak demand generated in the aggregate. Because all studios are not likely to be using all lines at the same time, this should be a lower number than the sum total of all lines appearing in all studios.

Generic PBXs will not do for our broadcast application – they just don't have the features necessary. For example, while lines may certainly be shared to multiple phones, there is no way to switch groups of lines from studio to studio. There is also no way to connect computers for call-screening applications. On the audio side, there is no adaptive hybrid or professional audio

outputs. Usually, there is only one or two “Music on Hold” inputs for the entire unit, while we need one for each studio. While you could use a PBX to derive analog lines for the studio telephone interface gear, it will be far superior to make a direct all-digital link. So we will need something like a PBX, but specialized for broadcast.

What we need for a modern broadcast studio telephone interface system is:

- Support for T1/E1, PRI ISDN, or BRI ISDN, with interface directly to Telco lines, or via PBX so lines can be shared with office phones
- High line capacity – perhaps 96 lines (four T1s)
- Multiple studios and producers, etc. can share the lines
- Program On Hold for each studio
- Users may switch line groups among the various studios according to need
- Caller ID
- User interface that is easy to learn and use, and minimizes errors
- Allows connection of computers over TCP/IP for call screening, production
- Remote management and diagnosis, over Internet
- High-quality audio and high-performance hybrids
- Digital connection to studio audio gear (AES/EBU or Ethernet IP)
- Ready for future, such as VOIP

The first point is clear: The system must plug without hassle to all of the usual modern digital Telco interfaces. You can choose to have the studio system to connect directly to the Telco lines, or via the PBX so that the lines may be shared with office phones. High-end PBXs support T1 or ISDN on the station side, so that can be used as a link between the systems. Because the path in the PBX is all-digital, there is no worry about audio degradation. With most PBXs, lines may be easily shared between the business and studio systems, including display of status and transferring from one to the other – something that has been difficult or impossible with older systems.

The ability to on the fly reroute set of phone numbers to different studios, as operations require, is an important requirement. The trend in larger consolidated facilities is 'generic rooms' used at different times of the day for different shows. Even in smaller plants, you may want to have a bank of call-in lines for a popular morning show that is re-used on a different station for requests at night, for example.

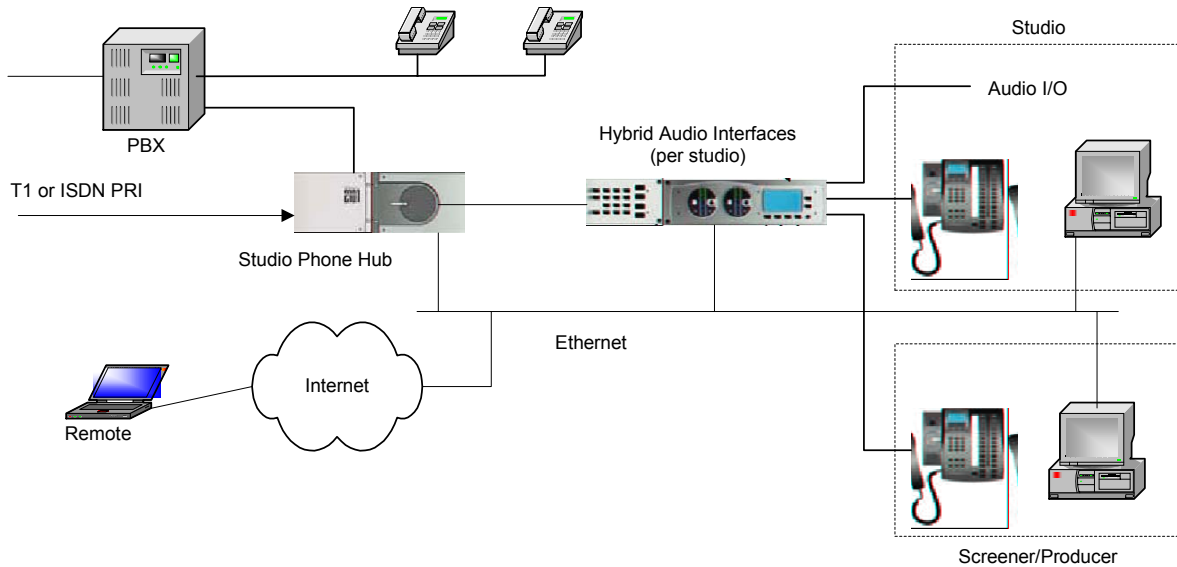


Fig. 3: Possible arrangement for a multi-studio system using T1 or ISDN PRI.

ISDN PRI inherently supports *Direct Inward Dialing* (DID), so you can have a separate DID "hunt" group for each show. Channels are not dedicated to specific numbers; rather, calls simply take the next available channel as they come in. The destination telephone number (the *Directory Number*) is contained in the associated set-up message so that the system knows where to send the call. If a morning show ends at 10 am and you call the morning show number you will get a busy rather than bothering the next show. Also, this allows you to have an unlimited number of hunt groups (only one per DID number, but many DID numbers are possible) and they remain under your control. You can do this with T1 also, but it requires that the Telco create the hunt group – a major loss of flexibility.

Program On Hold must be provided independently for each studio. It won't do to have callers to an AM talk show listen to the FM rock station while waiting to go on the air. It's a good idea to signal to the caller when the transition from Hold to On-Air happens with some kind of tone, as has been done with digital hybrids for analog lines for some time.

Caller ID is routinely provided with ISDN, both BRI and PRI. Unfortunately, you don't get this on T1. There should be a path for this information through the phone line interface hub to the screening computers via an Ethernet IP connection. With this, you could do database lookups to deliver information about caller characteristics, identify and screen problem callers, keep records from which to compile statistics, etc.

Remote management and diagnosis is becoming increasingly beneficial. A general trend is IP and Ethernet everywhere, and the system should support this. If the system is tied to the Internet, it would be possible to have

the customer support people at the manufacturer assist with any required configuration or troubleshooting.

Smooth Operations: Ergonomic User Interfaces

As with the PBX-like switching functions, the "phone set" used in a broadcast environment is similar to a common office phone, but surely not the same. Perhaps the most important difference is that the studio environment requires systems that help operators to achieve reliably smooth and error-free delivery of on-air elements. The studio phone system must be easy to use, presenting information in an unambiguous format, and allowing control without operators having to think through complicated button press sequences, etc.

Among the other differences and requirements:

- Display for line status needs to be clear and high-contrast, preferably iconic.
- Buttons need to be bigger and more widely spaced because talent needs to concentrate on performance, not the gear.
- A take Next ready line function can simplify sequential call answering.
- Busy All function can help to prepare for contests by clearing the lines.
- Two or more hybrid support for easy and high-quality conferencing.
- Should be able to move calls from handset to on-air hybrid, with simple easy to perform transition.
- Needs to work seamlessly and collaboratively with the screener and screening computer, when these are needed.

- Automatic control for recorder can be helpful when taking calls off the air.
- Control for special features needed in the studio environment such as moving DID groups from studio to studio are needed. These must be easy to use so that studio operators can make changes simply and without assistance. No complex sequence of entries.
- An option to mount it in audio mixing consoles is often desired.

Ergonomics research, which studies how humans interact with machines, has concluded that icons are the best way to indicate simple and repeated status changes to operators. Typical telephone sets use a single LED per line appearance to indicate the state, with different blink rates signaling a call ringing in, on hold, etc. This is an icon, but an impoverished one, to be sure. We can do better. Computer screens are tremendously richer, but also bigger and not contiguous with the selection pushbuttons. One solution is to use small LED matrixes to display the icons. Users can have the best of both –phone-like “desktop directors” can use high-contrast LED matrix displays, but computers connected via Ethernet, with large high-resolution monitors, may be used to view line state as well.

With a matrix display incorporated into the line director phones, it is possible to convey line information in a more sophisticated fashion than was possible in last-generation controllers with simple blinking LEDs. These Status Symbols can be used to indicate which lines are screened by the program producer and ready for air, for example.

A **Busy All** function makes all lines that are configured to be for public call-in go to a busy state. Normally, this is used to prepare for a contest to ensure that all callers have a fair chance to win the prize – and helping to avoid the litigation that has sometimes occurred when listeners believe they have been mistreated.

Conferencing is an important feature in studio phone systems. Standard PBXs require multiple button pushes in complicated sequence to make a conference connection. Because this is a common operation with call-in programs, this needs to be easily and flexibly accomplished. Good audio quality, with appropriate levels to all parties is essential for successful interaction. Multiple adaptive hybrids can be used for “main” and “conference” callers, such as is often required for shows with telephone guests combined with calls from the public. However, because digital telephone services are naturally “4-wire,” with independent send and receive paths, callers can usually be conferenced directly without hybrids. The downside is that you don’t have fader control over each caller.



Fig. 4: An LED matrix display in the studio director phone can convey line status information more clearly than the usual blinking LEDs. Some icons may be enhanced with animation.

Call Screening and Show Production

Call screening and communication of information related to callers between a show’s producer and talent is best done with computers. Thus, the system should support connection of computers via Ethernet.

Using Internet standard TCP/IP permits computers to be connected anywhere an Internet connection is available. It may also be possible for the talent to use a remote computer to select lines in the absence of a director phone.

Line Groups

Recall that DID groups are lines associated with a given telephone number. Support for these in the studio phone system enables you to walk into any room and on the fly choose for which station or program to take calls and produce in that room. You can even change rooms in the middle of a program, and calls on hold/screened hold can follow you.

Conclusion

Broadcast facilities being built in the year 2001 are very different from those that came before. Digital is everywhere, and computers and Ethernet are near universal. Many facilities are serving multiple stations and studio infrastructure is becoming increasingly sophisticated. Telco services, limited for decades to primitive 100-year-old analog, have at last made the leap to a modern digital form. Studio telephone interface systems that are designed for this environment are now ready to go. Installing one will improve audio quality, make operator's lives easier, and help engineers better serve the needs of their stations.

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