Telephone Technology in the Digital Age - A Tutorial for broadcasters

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On the Seventh of March, 1876 Alexander Graham Bell received a patent for the telephone. His patent was described as "Improvement in Telegraphy ..." The Bell Telephone Company was formed in July of 1877. While many at the time scoffed at this invention, and Bell had a hard time getting investment capital (the powerful Western Union refused his offer to sell his patent for $100,000 in late 1876/early 1877), the public loved the telephone and acceptance grew rapidly. By December 1877 Western Union had formed the American Speaking Telephone subsidiary to compete with Bell.

Bell's original apparatus was similar to a modern dynamic microphone. A permanent magnet and a piece of moving metal interacted to induce current in a coil (the magneto effect). The original system used the same transducer for both sending and receiving. At the far end the electrical energy was converted back to vibrations thereby creating acoustic waves (i.e. sound). As early as 1878 telephones had become available in places as far west as San Francisco with the first exchange installed there in the January of that year.

Transducer development

Once it became clear that the telephone was, to use a modern term, a "killer app," a number of other parties came up with alternative methods of telephony in order circumvent Bell's patents. A lot of energy and inventiveness (not to mention money) were invested in "wire" telephones which work on the same principle as two cans connected with a piece of string (or wire), since this was clearly a legitimate way to beat Bell's patent. Indeed, patents were even issued for **exchanges** for this technology! Inventor Thomas Edison (who had yet to invent the phonograph & electric light) and others were contracted by Western Union to develop new and improved telephonic technology. Edison invented the carbon transmitter (or microphone) which became widely used for telephonic applications until the mid 1980's. The carbon microphone works on the principle that carbon granules that are loosely packed will vary widely in resistance when disturbed. By impacting the carbon granules using a mechanical connection to an acoustic diaphragm, the resistance becomes proportional to the acoustic energy hitting the diaphragm. A battery was used to provide a source of current (note that the original Bell technology was "talk powered"). This technology worked well as a transmitter, but was entirely inappropriate as a receiver. When coupled with one of the Bell transducers this transmitter worked better (higher levels and less distortion) than using two of the Bell devices.

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The infant telephone industry grew rapidly. While the Bell Company had a head start, the giant Western Union rapidly caught up. Eventually, the courts determined that Bell's patent had a rather wide scope and a settlement between the two companies was made in November of 1879. The Bell Company got the rights to the Edison transmitter technology, along with a network of 56,000 telephones! Western Union agreed to stay out of the telephone business with the Bell Company paying them 20% of telephone rental fees until the Bell patents expired. Two years later Bell bought Western Union's telephone manufacturing division, Western Electric. It became the sole supplier of equipment to the Bell System until divestiture in 1984. Lucent Technologies is the direct descendant of that company.

The standard transducers used became the Edison carbon style transmitter and a Bell style receiver. Note that there were numerous improvements along the way, however this basic technology was used for nearly all telephones right up to the 1980s. Many such phones are still in service today. Also note that the telephone required batteries (originally large wet cells, later the more compact and less dangerous dry cells were used) until conversion to "common battery" system beginning in the 1890's.

Signaling - Alerting

The original telephones, as mentioned above, each had a single transducer which the user moved back and forth from mouth to ear to communicate. The typical arrangement was two or three telephones directly wired together. However it became evident quite early that there needed to be some way to alert the far end that someone desired to talk. The earliest approach was to yell into the phone and hope someone heard you at the far end! The first technological solution to the "call alert signaling" requirement was to tap the telephone transmitter with a pencil or a small hammer. Once phones began to have separate transmitters (mounted in a wooden box) and receivers (mounted in a wooden or rubber hand-piece) it became possible to build a "thumper" into the box with the transmitter. This allowed the user to pull a knob and a hammer would strike the diaphragm of the transmitter.

Very soon thereafter Thomas Watson, Mr. Bell's original assistant, came up with the idea of using a hand-cranked magneto to create an AC voltage on the line couple with a simple electromechanical ringer at the far end for signaling. A condenser (capacitor) served to isolate the ringer coil from the DC battery circuit. This technology already existed at the time, however its application to the telephone was much needed and worked well. Users could hear if someone wanted to speak to them on the telephone, even if they were not right near it at the time.

Switching - Early manual exchanges

As mentioned above, the early telephones lines were generally two or three phones wired in parallel. For instance (to use a dated example appropriate to those times), an affluent family might have a telephone from the home to the husband's place of business. And possibly another phone from the home to the local store so the wife could call and request that provisions be delivered. It became rapidly clear that the utility of the telephone was directly proportional to the number of people to whom one could talk. For instance, to use our example above: If the businessman were to call and inform his wife that he would be bringing home an Associate for dinner, the wife could then phone the store and get more provisions. If the residence had only one line (to either location) this would not be possible.

However, the practical matter of many individual lines running to many other locations was problematic. If the system were to work as desired, every time a new phone were added lines would be required to each existing phone. Also, each time another phone was bridged onto a line the level dropped. So these would need to be separate lines, possibly with a phone on each end.

The solution to this dilemma was the Central Telephone Exchange. Instead of running lines from subscriber to subscriber, each line went from the subscriber's location to a central location, the telephone exchange. Here the lines were terminated on a "switchboard". The first such telephone switchboard was installed in Hartford CT in 1878 with 21 subscribers.

Switchboard technology evolved rapidly. The typical switchboard worked as follows. Each subscriber line came into a "jack" (originally a "jackknife switch", but the term remains). An electro mechanic device called a "drop" bridged the line. When the subscriber cranked their magneto the drop released a small metal hinge indicating to the Operator at the Central Exchange that the customer was requesting service. The Operator could then plug into that jack and ask the customer what they desired. The Operator could then ring the requested party (using a hand magneto at first, later large "ringing machines" driven by water wheels or a steam engines were used to provide ringing current). If the called party answered, the Operator would proceed to connect the two parties together using a "cord" or connecting cable. The operator would then reset the drop. When the parties were finished they would ring their magnetos and the operator would come back on line to check to see if they were finished. If so, the connection would be "pulled down" by unplugging the cable.

Note that there is a second signaling function implicit in the above arrangement. The calling party would tell the Operator to whom s/he desired to be connected. This signaling function is an example of "in-band signaling". In other words, the "signaling" occurs within the same channel as the communication itself.

Later, in 1891 the automatic exchange (**switch**) was developed. The system rapidly evolved to where a rotating dial pulsed out DC pulses that then drove a series of stepper switches of one kind or another. These dial "pulses" are another form of "in-band signaling". The automatic system was adopted fairly early by the **Independent** Telco's, but was not widely adopted by the "Bell System" until much later in the 1920's through the 1940's.

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The economics of telephony

Central exchanges had a number of important advantages for the telephone companies:

1) Only one pair of wires was required to each subscriber's premises.

2) Simplified plant requirements; The Telco could run extra cables from the central office to the vicinity of a growing area in advance. The designers knew in advance that all circuits will terminate at the central telephone exchange (now often called a Central Office). They could therefore have these circuits pre-installed and terminated at a central frame ready to be hooked up to the switching equipment prior to receiving a call for new service.

3) They could build and share "toll" circuits and other facilities among users. For example, one or two toll circuits (called toll **trunks**) could be built from town A to town B. Any subscriber in either town could take advantage of this facility, most likely there would be an added "toll charge" for this call. The cost of building the trunk was thereby shared among all users who had need to call between the two towns. Even if a wealthy subscriber had been able to afford a line between the two towns, under the pre-exchange system s/he would have been limited to speaking to a single location at the other end. And no one else would have had access to this facility without disturbing this individual.

An important principle soon became evident when building telephone exchanges. While many people might have telephones, at any given time only a few are actually using the telephone. So the number of "paths" through the switchboard (i.e. the number of Operators and the number of cords) can be far lower than the number of lines coming into the switch. The science of Traffic Engineering attempts to determine the number of paths required to keep the amount of "**blocking**" (inability to complete a given call at a given time) to an acceptable level. It applies to both **trunks** and to **switch** paths. There is no simple solution to this problem as economic issues must be considered as well. Each trunk added between Town A and Town B will reduce **blocking**. However, the more trunks put in, the greater the number which will be unused except during periods of peak demand. The inevitable result is that in order to maintain economical service *some* blocking will occur ("all circuits are busy... please hang up and try again later..."). The state regulators determine level of blocking that is acceptable and the Telco must meet that level or risk being punished.

Theses principles are important to understand as they determine the very basis of economical telephony. Shared facilities are essential to economical operation. Note that the same principles apply to private telephone systems. However in this case *you get to decide* what level of **blocking** is acceptable rather than the regulators. As you add more trunks to your **PBX** you will reduce the chances of users being unable to get an "outside line", however you will need to pay for the **trunks** every month. And a bigger **switch** will allow more simultaneous paths through the switch, however a bigger switch costs more money.

There are a few disadvantages to the central exchange method as well:

1) If you desire to talk to your neighbor, the audio must still run from your location all the way to the **exchange** and from there back to your neighbor's location. This method can have a significant effect on signal level and noise floor.

Even should you purchase a dedicated line from your facility to a neighbor's location, the Telco infrastructure will still mandate that the line go through the **central office**. Readers who have dealt with high fidelity conditioned audio loops are well aware of the frustration involved with having your audio routed many extra miles with the resulting performance and reliability limitations

2) Some **blocking** is inevitable

3) Single point of failure. A fire or equipment failure at the **central office** will affect multiple users.

An introduction to Private Telephone Systems

Many readers will recall seeing old motion pictures where multiple phones are shown on a worker's desk at a business establishment. Film writers often took advantage of the absurdity of talking on more than one phone at a time, or trying to determine which of several phones was ringing, for comic effect. However there was a time when this was exactly how telephones were used in the office. Two different approaches were developed to solve these problems, both of which still exist today. **Key telephone systems** and the Private Branch Exchanges (**PBX**).

Key Telephone Systems

The **key telephone system** is a direct evolution from having multiple phones on the desk. The earliest system had a series of "keys" (switches) mounted in a box, which allowed one to choose which of several lines was connected to the phone. One position of the key "hung up" the line while another connected it to the phone. Usually an intermediate position of the switch allowed one to place a call on "hold". However there was still the problem that if more than one person was sharing these lines that each person would not know if a line was in use without checking it, thereby interrupting anyone who might be using it. In addition, callers could be forgotten on hold. Not only did this annoy the customer, but that line would not be available for use until the mistake was discovered. "Supervision" between stations showing the status of lines was required.

True **key telephone systems** have a Key Switching Unit that all lines are connected to. Standard **loop start lines** are normally used. All phones in the system connect to this KSU. Each phone in a Key system typically has access to several lines, *through the KSU*. The phones and KSU work together as follows. Each phone has an indicator for each line to which it has access. These indicators allow the user to see that state of each line to which they have access. The for instance, if a line is ringing, the user can determine which line(s) it is by looking at the indicator. Or if a user desires to dial out s/he can select and idle line by looking at the indicators and selecting a line shown as idle. Once the line is selected and the handset lifted it is as if this phone is actually connected to the line itself. However, the KSU will provide supervision so that other users are aware of the status of this line. In fact, with the early "mechanical" key system such as the 1A1 and 1A2 the user's phone is actually connected to the line through a "hard" connection (switch contacts) when a line is in use.

The distinctive characteristics of a Key Telephone system are as follows:

1) Each **line** on the phone has a direct correspondence to actual lines from the Telco. In older key systems such as the 1A1 and 1A2 six conductors for each line go to each phone on which it appears!

2) A given line can, and often does, appear on more than one telephone set. Which lines appear on which telephones is determined when the system is installed. This relationship is more-or-less permanent.

3) The system communicates line status to the users who supply the "intelligence" of the system. It is up to the user to select an unused line before dialing out. If s/he does not s/he may interrupt a conversation in progress or inadvertently answer a call ringing in. Likewise, if a line is ringing it is up to the user to select the ringing line before answering.

4) There is no way to directly "call" from one phone in the system to another. Any number of users may select an "intercom line" (a common circuit with just talk battery on it) and talk, however this does not go through the KSU or really constitute a "call" (ie a switch connection). Or one user may call out on one line to another line of the system, and hope that the desired party answers the ringing line. For these reasons a public address system is often used with a Key system to allow announcements to users that they have a call on a certain line.

To summarize: Key systems allow multiple phones to efficiently share phone company lines. Each line has an identity (the phone number) but the telephones do not.

Of course modern Key Systems often offer features that tend obscure some of these characteristics. For instance, modern key systems may only require 2 or 4 conductors to each phone. Some modern systems may automatically select a ringing line, etc.

Private Branch Exchanges (PBXs)

A private branch exchange is just like a telephone company exchange. It has one or more "telco **trunks**" to the phone company. And it has one or more "**station lines**" to telephone termiunals. Just as with telephone company exchanges, a switchboard and Operator were originally used to place calls. Later, Private Automated Branch Exchanges were developed where direct dialing out of the system became possible without an Operator.

With traditional **PBX**s an attendant (Operator) was also required to answer all incoming calls and route them to the intended party.

Key characteristics of a PBX

1) Each phone has its own "station line" from the **PBX**. While sometimes phones will have more than one line, never does more than one phone share the same line.

2) Inside calls between stations of the PBX are possible without going through the telephone company **trunks**.

3) **Trunks** (to the Telco **switch**) are shared between all users. To make an "outside" call the users dials a special digit (typically a "9" or "8"). If the **PBX** has an available **trunk** the users is connected it. The user then hears or "draws" dial tone from the Telco switch and can proceed to dial the number. Unlike a Key system the user, has no indication until after s/he picks up the phone and dials the special digit if a trunk is available.

To summarize; **PBX** telephones have an identity of their own. They are extensions with a unique extension number. Each has access to the PBX. The PBX has trunks to the phone company. The PBX actually switches calls through itself based on the users demands, not based on a pre-configured wiring plan. Station-to-station calls as well as station-to-trunk calls are possible.

Modern business telephone systems can still generally be categorized as Key systems or PBXs. Note that the advanced features available on many modern Key systems can blur the difference unless one looks carefully.

Centrex

In the 1970s the deregulation of the telephone companies began and end users began to have the right to purchase telephone systems from suppliers other than the AT&T owned phone companies. The end user suddenly had many more choices. To compete with the PBX and Key systems, now available from outside vendors, the phone companies began to offer another option called "Centrex" service. This option became the only option the phone companies were permitted to offer once the divestiture of AT&T was complete in 1984. Only since the 1996 Telecommunications Act has been implemented have the major Telcos been permitted to provide equipment at the user's premises such as Key and PBX equipment. Therefore Centrex was an important product for the telephone companies during the 1980's through 1990's.

Centrex is really very simple. Each user has a phone, and each phone has a line from the phone company and a telephone number. However, in order to meet the needs of businesses, Centrex lines have a number of important features. Centrex is designed to act much like a PBX. To dial another person in your Centrex Group you need only dial 4 digits (in some arrangements 3 digits). These intra-Centrex calls do not occur any per-call or per-minute charges. An unlimited number of these calls are included as part of the Centrex package.

Other Centrex features include:

- Transfer to another Centrex line
- Hold
- Busy/no answer forwarding to another Centrex line
- User specified call forwarding
- Call pick up (ability to pick up a call ringing on another Centrex line).
- Hunt groups
- Conferencing

One feature that is usually available on Key and PBX systems is generally not part of Centrex, music on hold. This feature is sometimes available, but is complex to order and install since the audio must be transported to the telephone company **central office**.

Centrex is sometimes used as a way to bridge between two systems, For instance, if your on-air and business systems both used Centrex you could transfer calls from one system to the other using the Centrex transfer function.

The major advantage to Centrex for most users is the convenience of "one stop shopping".

The single biggest drawback to Centrex is the fact that it is typically used with basic telephones. The user must "hookflash" the line (hang up the line for less than 0.9 sec) and then enter a string of digits to request the above features. It is confusing for the users, and the proper codes for a given feature are easily forgotten. Recently a number of "smart phones" have become available for use with Centrex. These phones have feature buttons to make things easier for the user, although the actual signaling is conducted using loop current interruptions and **DTMF** tones.

Digital Phone Systems and the broadcaster.

Introduction, early digital systems

In the 1970-80's both **Key systems** and **PBX** systems began to offer systems where the internal workings of the system (and often the link to the proprietary telephone sets as well) worked with digitized audio. By using digital audio paths these systems could offer more advanced features (such as high quality conferencing) and better prices. Part of the drive to do so was the availability of low cost digital components due to the surge in popularity of the computer and related digital technologies.

The switch to digital systems was not always embraced by those in the broadcast industry for several reasons. Once DSP (digital signal processing) became available, the adaptive hybrid became possible, allowing true full duplex on-air audio without corruption of the “announcer” audio. When Steve Church (our founder) designed the first adaptive DSP hybrid, the Telos 10, he designed it to work with the 1A2 Key systems that were ubiquitous at the time. It could also be easily adapted to direct connection to a single phone line or a home-built line switcher.

The new digital phone systems required a special “analog port” adapter to talk to any analog device (including faxes and modems), which complicated matters. More importantly, it was found that the Analog to Digital to Analog conversions of the signal as it passed through the system severely degraded hybrid performance. One potential solution, to connect to the system digitally, was not feasible since the digital phone ports were proprietary. Manufacturers would not release details, and each used their own scheme.

Therefore the best approach for many years was to use direct lines from the phone company.

Digital trunking

As businesses began to buy digital switches, the practice of bringing digital circuits to the user’s premises from the Telco became more practical. While very high volume users had sometimes used digital trunks in the past, often converting them to analog for use with analog PBXs or other equipment, the newer digital PBX’s could frequently support direct connections to these digital trunks. Usually T1 Telco circuits are used for digital trunking purposes, although PRI is another option.

The advent of the digital **PBX** connected directly to a digital **trunk**, combined with the demand for better quality analog ports by modem users, has made it practical to operate high quality adaptive hybrids off many PBX systems. Since the network is nearly 100% digital today, this means that the caller’s voice is digitized at his/her telephone company central office and transmitted digitally all the way into the PBX. There, it is converted back to analog. If the analog port is of good quality the small amount of analog cabling between the PBX and the hybrid is insignificant and very good hybrid performance usually results. This arrangement is well suited to situations where a single hybrid or pair of hybrids is used for interviews or production purposes.

ISDN PRI is the ideal solution for flexibility. The **D channel** protocol allows for sophisticated operation difficult to achieve in any other way. Incoming calls requests are delivered to the **PBX** which can then accept or reject based on its own requirements. When a call is accepted, the PBX determines which channel to accept it on. This allows the same sort of efficiency enjoyed by telephone company trunking arrangements for quite some years. No **channel** is used just to return a busy signal.

DID

Recall that telephones on a **PBX** have an identity of their own. It is possible to have the identity of a PBX phone be a "real" telephone number rather than just an extension number. A block of numbers can be purchased from the Telco for this purpose. When combined with a special "**DID**" (Direct Inward Dial) **trunk**, a specific extension of a PBX can be dialed by users outside the system. Traditionally DID trunks could only be used for calls inbound to the PBX (hence the word "inward" in the name) which limited their affordability to large PBX owners only. With the advent of digital trunking "two-way-DID" has become available. Using **T1** or **PRI** these trunks can handle both inbound and outbound traffic and still have the ability for outside users to dial specific extensions as desired.

Buying dial tone from a phone company other than your ILEC (Incumbent Local Exchange Carrier)

Under the provisions of the 1996 Telecommunications act, the local dial tone business in the USA has become much more competitive. Deregulation is the trend throughout the world. This author believes this is a good thing and that competition will benefit all, even those who stay with the traditional provider. The following are some questions you should ask as part of deciding if you should change providers:

1) Does the provider merely resell the services of some other company? Do they own their own **switch**ing equipment?

2) Does the provider own **trunk**ing to your neighborhood, or will they be leasing copper/fiber from a **ILEC**?

3) What are the charges (per minute and/or per call) for local calls? It is likely these charges will be structured somewhat differently from your existing carrier. Scrutinize your usage (if you have a PBX ask your vendor to generate usage statistic reports for a couple of months) and calculate what your actual charges will be under the new plan.

4) What duration is their minimum contract? Do longer term contracts offer a meaningful discount?

5) Does the provider offer a 24 hour repair hotline? Most incumbents do not. Will they give you the number direct to their switch technicians?

6) Contact some references and see how satisfied they are with the customer service and company in general.

Our experience is that the new **CLEC**s (Competitive Local Exchange Carriers) are often more "friendly" to do business with. Often, their smaller bureaucracies allow you a better chance to get through to the person you need to deal with easily. Employees often appear to care about you as a customer. However, there can be a drawbacks too; these leaner companies have less redundancy and may have only one expert in a given specialty in your region to rely on. And, if they are leasing local loop facilities from the ILEC, they may be slower to get things installed (or repaired) as they must rely on the (sometimes unwilling) cooperation of the ILEC.

Remember, that in today's competitive marketplace you may very well have the ability to negotiate with the vendor. This may even be true with the ILEC.

A few words on Long Distance Providers (Interexchange Carriers)

It has been over 15 years since basic long distance services were deregulated. Just as with the choice of dial tone providers, you should choose a long distance provider based on something other than just price. Most stations would not buy the cheapest components for the air chain. Remember that the quality of your caller audio can reflect on your reputation.

1) Does the provider merely just resell the services of some other company? Do they own their own network?

2) Does this long distance provider offer "**Circuit Switched Data**" connectivity? If you are making outbound long distance calls with a codec (such as the Zephyr or ZephyrExpress) this is required. Be aware that most do not. If in doubt get their 10xxxxx dial-around code (in the USA) and try some calls before changing carriers. Get the number for their "switched data service center" who can troubleshoot problems with switched data calls. You may decide to change your carrier for all lines except the lines used for your codecs.

3) Is your carrier using any form of "voice compression" on calls? If so, audio quality and hybrid nulling may be unsatisfactory. Again, if in doubt, ask for the dial-around codes and try some calls.

4) Look at both per minute as well as fixed monthly charges when comparing rates. You will need to determine your current usage to calculate what your actual charges will be.

5) Consider international long distance needs carefully. If you do not need it you may wish to get a plan that does not allow international dialing. If you need it you must be sure that the rate for international calls is also reasonable. Again, using your actual past history is the best way to make informed judgments.

5) Be wary of long-term contracts. Rates continue to fall regularly. You should be checking the marketplace every 18 months or so and renegotiating rates accordingly. Negotiate your cancellation charges as well as your per-minute rates.

Always be sure to go back to your current provider and see if they are willing to renegotiate their rates. This is a particularly good strategy if your main motive for changing is to get lower rates and your current service is working well.

6) Contact some references and see how satisfied they are with the prospective company and their customer service.

Broadcast On air requirements for telephone systems

On air telephone systems require the ability to “stack up” callers who can then be put on the air as needed to maintain the continuity and artistic needs of the show. Easy access to each caller is essential to allow the show to progress smoothly. The “Key” approach works best here because multiple callers can be accessed with certainty and immediacy. In addition, call screener systems (both manual and computerized) can refer to a caller on a particular “line” and be understood. Therefore most broadcast telephone systems are either based on a key system such as the 1A2 (preferred because it is easy to use, non proprietary, and offers “hard” connections rather than using electronic switching of the audio path) or a specialized self-contained system that follows the “Key” style of operation (such as The Telos One-x-Six, DIM or TWOx12).

When a digital **PBX** with digital **trunks** is available, we have frequently seen a number of analog ports fed from the PBX to a 1A2 key system, or directly to one of the self contained Telos systems mentioned above. This approach generally works quite well. However, with the Telos TWOx12 digital version several new all-digital options are now available and offer superior performance.

Behind the PBX

Many of the current generation of **PBX**s can support **ISDN BRI** terminals. Some offer the 2 wire **U interface** while others offer the 4 **wire S interface** (USA & Canadian users note; if you require a TWOx12 with the S interface you will need to special order the version normally shipped only to outside the USA & Canada. Your Telos Dealer should contact Telos for ordering information). While the physical interfaces are well defined, there are some issues regarding protocol compatibility. You should consult the list elsewhere in this manual for PBXs which have been tested with the TWOx12.

The “behind the PBX approach” has a number of important benefits beyond the benefits of a fully digital path from the caller’s central office:

1) When more than a few **trunks** are required, digital trunks frequently cost less than analog trunks. By combining your on-air and business-related telephone traffic you are better able to take advantage of these economies.

In addition, you may choose to have separate digital trunks going to your long distance carrier. This permits substantial savings on your long distance bill. If the same long distance also handles your inbound toll free lines for on-air use, you should be able to achieve savings on your inbound long distance rates as well.

2) Callers to the main **PBX** number(s) can usually be transferred to the on-air system just as they can be transferred to any other extension in the system. Note that transfers from the TWOx12 back to the main system may not be possible.

3) Caller ID is more likely to be supported on a PBX BRI port than a PBX analog port.

4) Using **DID** (Direct Inward Dialing) numbers and hunt groups on the station side of the PBX, you can gain considerable flexibility and control vs. a Telco hunt group on analog **trunk**s.

For instance, by using the "called number" information available on incoming calls the PBX could route calls to one of multiple inside hunt groups. Instead of ordering a separate group of lines for your Morning Show and syndicated Talk Show you can share the digital trunks from the Telco and just route the calls within the PBX appropriately.

Bringing it all together

New standards-based approaches to digital telephony such as ISDN BRI & PRI offer substantial new opportunities for the broadcaster. When selecting a PBX for your broadcast facility you should be sure to investigate the following:

1) Can the system support **ISDN BRI** or **PRI** (system 2101 only) on the user side (extension side) of the switch? If so, is this standards based, or proprietary? Are both **circuit-switch-data** and **circuit-switched-voice** connections possible from these extensions? What **ISDN protocol** family is used?

2) Does the system support digital Telco trunking (i.e. **T1**, **E1**, **PRI**)? Can the system support **two-way-DID** over digital trunks? Can a **hunt group** of extensions be accessed using a single DID number? Can the same hunt group be accessed using a second DID number?

3) Can station personnel readily change hunt group configurations? Add new stations? Is this level of self management included in the training package or is this extra? This is important not only as your station (or group) evolves, but as a tool to be used for special events or as part of your disaster recovery plans.

4) Can multiple **BRI**'s be in a **hunt** group?

4) Can a user on a "feature" telephone transfer a call to a BRI/PRI extension? Can a user on an analog port transfer a call to other extensions on the system using a hook flash transfer?

5) Are high quality analog ports available for the system? Do they support high speed data (33.6 or 54 Kbps)?

We hope this tutorial has served to make you aware of the issues involved in selecting and managing a telephone system for broadcast use in the 2000's. While it is likely that there is more than one solution that will adequately meet your needs, we hope that this discussion will help you build a system that meets your needs and also takes advantage of the economies possible when the newer technology is applied to today's broadcast facility.