APPLICATIONS

AUDIO ON DIAL-UP CIRCUITS

DIAL-UP AUDIO

▼ 15kHz full duplex audio on a telephone line - the Comrex Vector

DESCRIPTION

Live remote audio links are increasingly important in radio broadcasting, particularly news and sport, and the traditional methods of establishing a link have all but gone. Broadcasters have turned to dial-up networks as an alternative, and often much less expensive, route to the studio.

There are several major dial-up networks, each overlapping one-another and each providing a particular service relevant to the broadcaster. By dial-up service we mean one that establishes a link only when required, thus lowering costs and, is broad-based, ensuring that availability is widespread.

The best-known service is the public switched telephone network (PSTN). Many devices have been developed to get more out of a telephone line, and current systems are extraordinarily good.

The Integrated Services Digital Network (ISDN) is the digital equivalent of PSTN, capable of giving a very reliable service where it is available, with various products offering high-quality audio links.

Mobile networks offer great freedom in choice of location and, whilst they are no use for broad-band signals today, they promise the capability of carrying broadcast audio and more in the future.

Satellite uplinks offer the greatest flexibility in location, being available worldwide, but there is a significant premium in terms of capital outlay, call charges and the relatively bulky equipment involved.

IP services via the internet are mainly used by broadcasters for file transfer, but, as service levels improve and guaranteed routings become available, live links are increasingly a worthwhile option.

PSTN

▼ The PSTN has been used for decades to connect a remote reporter to the studio, and many products have sought to make the process simpler and the audio quality better and overcome the limitations of a two-wire system.

Telephone couplers simply provided a method of attaching audio equipment to a telephone line while protecting each from the other.

Telephone hybrids offered an acceptable level of separation between incoming and outgoing audio and allowed professional audio equipment to connect directly to the network.

Frequency extenders adjusted the frequency range of audio as it went into the network and came out at the far end. The adjustment allowed the resonant lower frequencies to fit into the narrow bandwidth of conventional telephone communications, losing only a small fraction of the higher frequencies as a result. More sophisticated extenders provided a broader range, utilising two or three telephone lines.

Now, telephone codecs offer very high quality audio, in both directions, over a single line. These convert analogue audio into data, use processors to reduce the volume of data without losing much audio quality, and then use a built-in modem to send this data via the line. At the far end, an identical device reverses the process. The delay caused by audio processing is relatively short and most products are suitable for live links.

Currently, the telephone codec offers the best audio quality obtainable on a phone line. Given the widespread accessibility to phone lines, it is possible to make a connection from almost any inhabited place in the world. Some phone lines will be too poor to make the connection worthwhile, but in general good quality audio can be sent and received by telephone codecs.

▼ PSTN is used mainly for news and sports reporting, the relatively short audio processing delay allowing seamless duplex links

▼ ISDN services provide the full range of broadcast links, from mono speech to stereo music, with equipment designed specifically for each application

ISDN

▼ ISDN simply takes the current telephone network and extends the digital capabilities from the exchange to the user. Three types of service are available: voice (basically the same as conventional telephones); asynchronous data and synchronous data.

The two data services are used to transmit high-quality audio over ISDN. Asynchronous data is used when sending audio from/to devices such as computers, as transmission can be paused while one or other device is busy doing something else. Typically the duration of the call is longer than the audio content, rendering asynchronous calls unsuitable for live transmissions.. For wideband stereo the call may take several times the length of the audio.

Synchronous data is always transmitted in real time, albeit with a short delay for the processing. An ISDN codec processes digital audio to reduce the volume of data to manageable proportions, and sends this data via a terminal adapter (often built into the codec) over ISDN. ISDN codecs are usually full-duplex, sending and receiving audio simultaneously.

The ISDN service is presented in 64kbits/second channels, called bearers, and several bearers can be combined to provide a greater throughput. This process may be performed by the terminal adapters at each end of the line, or by the codecs themselves.

Various different data reduction processes are used, primarily trading speed of processing against audio bandwidth. Codecs that reduce audio data by a factor of four are generally very fast, while those which reduce data by a factor of twelve or more can be too slow for interactive work. In many cases, the latter use the return audio channel simply for communications.

Mobile Networks

▼ **Current digital mobile networks offer asynchronous data links at very low bitrates, hardly suitable for broadcast applications. The next generation (third generation or UMTS) of mobile services promise data links of up to 2Mbits/second, offering the possibility of broadcast audio links from anywhere covered by a cellular network**.

Network providers hope to have services in the UK in 2002, and many European countries will not be far behind. Legislation will support roaming, overcoming the short-term problem of limited geographical coverage in the early days.

Services have yet to be agreed, data channels as wide as 2Mbits/second may prove expensive, and the asynchronous system may not be ideal for broadcast services, but the freedom offered by being able to just dial and connect without searching for a suitable wall-socket is an extremely valuable resource.

▼ Mobile networks offer total portability within their service area, and 3G networks promise high quality audio links

Satellite Phones

▼ **Satellite data services provide high speed digital data links from anywhere in the world, offering a service similar to ISDN without the need for an ISDN socket.**

Satellite services such as Inmarsat-B provide a 64kbits/second synchronous data link similar to ISDN, and many ISDN codecs may be freely used with these services. The typical concerns are that of power, size and weight. Even when a satellite uplink is described as portable, it is still a sizeable thing and any additional weight must be carefully considered.

High costs, both capital and call charges, tend to limit the use of satellite phones to newsgathering in areas without a basic communications infrastructure.

▼ Satellite phones are the established way of getting news from trouble-spots, as the user is not reliant on any local services

IP Services

▼ **Internet Protocol is the process by which the packets of data which make up web pages and files are sent from one address to another. This is not strictly a dial-up circuit, as packets of data can be sent by varied routes, arriving in any order at the destination.**

Using the internet to establish a broadcast link to the studio has typically meant sending an audio file as an attachment to an e-mail. As with PSTN and ISDN services, the audio information is often reduced in volume to provide, in this case, shorter messages. Audio processing can be achieved by a program running on a PC or by a dedicated hardware device with a PC-like interface to the internet.

Live audio links are limited by the data bandwidth available along the route. In general the user has little control over this at any point beyond the short hop to the service provider, and no control over the route the data follows. Often, routes are determined by current usage and by traffic agreements between service providers. For example, current VoIP services (internet “telephones”) are aimed squarely at cost saving rather than audio quality.

▼ IP services are ideal for off-line links, and service level guarantees may make this medium suitable for live contributions in the future

Access to the internet is getting faster at every stage of a packet’s journey, and service providers are starting to offer service level agreements for guaranteed transmission of data by a particular route or within a particular time. While these have to be arranged in advance they will be of limited use to the broadcaster, but the opportunity for live, duplex audio over the internet is clearly increasing.

AUDIO ON DIAL-UP CIRCUITS

All dial-up services pose certain challenges for the broadcaster. Whilst offering less expensive and much more extensive access, this comes at the cost of sharing the network with others. Being general purpose, sometimes the demands of broadcast traffic exceed the normal use of the network, giving poor results. As ever, there are compatibility issues, too. Here are a few hints regarding both choosing and using equipment on dial-up services.

PSTN networks are usually very good in cities but vary greatly in quality across the world, and the caller has little control over the route a call takes. Additionally, a wall socket which purports to be a direct exchange line can sometimes be quite different, perhaps part of a local digital exchange in a hotel, or a party line with an extension in the next room. In looking for a codec for the PSTN network, several considerations beyond bandwidth and processing delay are important. A low connect rate at which useable audio is provided, maintaining a short delay at lower connect rates, an option to determine the rate used, options for dealing with noisy lines, failsafe connection when the line won't support any data link, international telecom support, an alternative method of dialling (in case standard tones are not recognised) are all important facilities. The Comrex Vector provides all these facilities and more and, as you would expect from a company which has been dealing with telephone lines for so many years, the Vector comes with plenty of information on getting the best connection possible.

ISDN is a very reliable service but is less widespread, and it either works or you get nothing at all. Protocols for the connection of equipment vary across the world, and some terminal equipment cannot be used with certain ISDN services. Clearly there is a need to consider a backup service, perhaps a PSTN codec. The most important consideration for an ISDN-based service is the choice of algorithm, which determines the audio bandwidth, mono/stereo operation, and the processing delay. G.722 is the established favourite for mono speech links; MPEG for most applications requiring wide bandwidth and/or stereo, such as concert relays; and apt-X for STL backups and for music links where the delay must be kept to a minimum. Despite sharing a common algorithm, not all equipment is fully compatible, so tests with the actual equipment to be used are vital. The Vortex Baby Blue neatly bridges the need for a quality contribution and a talkback without delay, offering asymmetric connections.

The mobile phone service is not a player in the broadcast audio field, currently, and as the definition of future services has yet to be decided there is little relevant advice to offer at present.

Satellite links are well-established and reliable, but owing to their cost and the cost and weight of the equipment they tend to be used for news-gathering in areas where there is no reliable PSTN or ISDN infrastructure. When a codec is used to give wideband audio, it tends to use the G.722 algorithm and the compact portable models such as the Comrex DXP.1 obviously find favour.

Internet Protocol is ideal for audio file transfers, being accessible in most locations and offering local call charges. Current service levels make it unsuitable for live audio of any quality, but service providers are beginning to offer guarantees of throughput and also control over the route (and therefore the number of nodes through which the signal passes), allowing the delay through the network to be reduced. Widespread availability of ISDN access is improving matters, and the advent of premium access via *X*DSL (digital subscriber line) services is also increasing accessibility. Whether the delay becomes short enough, and the throughput reliable enough, for live audio remains to be seen.

Vortex Communications Ltd
75 The Grove, Ealing, London W5 5LL, UK
Tel: (+44) 020 - 8579 2743 ▼ Fax: (+44) 020 - 8840 0018
E-mail: info@vtx.co.uk ▼ Web: www.vtx.co.uk

VORTEX
Communications Ltd