IP Broadcasting & the Importance of Session Initiation Protocol

Broadcasting over IP is rapidly becoming the paradigm by which broadcasters are planning future broadcast network infrastructures. Within the diverse range of broadcast IP devices coming onto the market, Session Initiation Protocol (SIP) is currently the signaling protocol that is used by most of the world's Telcos and broadcast codec manufacturers. It is also the most likely to provide connectivity between IP devices for the foreseeable future.

To understand the context in which SIP is developing as a signaling protocol, it is useful to know some background about SIP, its history and how it is currently used in broadcast products.

What is SIP

Session Initiation Protocol, or SIP as it is known, is an open standard application layer signaling protocol used to connect and disconnect connections over IP networks and the Internet.

SIP can work with a myriad of other protocols to establish connections between all sorts of different devices and it is capable of supporting audio, video and instant messaging technologies, without regard for the particular device or the media content that is delivered.

Historically, it has been widely used in VoIP applications. It has also been used for multimedia distribution and multimedia conferencing and can be used to create two-party, multiparty or multicast sessions.

More recently, its ability to create and manage connections over the Internet has led to its integration into an increasing number of broadcast audio and video products that stream data packets in broadcast applications.

How does SIP work?

There are two distinct parts to a call when dialing and broadcasting over IP. SIP is used in the initial stage for call setup. The second stage is when data transference occurs and this is left to the other protocols used by a device (i.e. using UDP to send audio data).

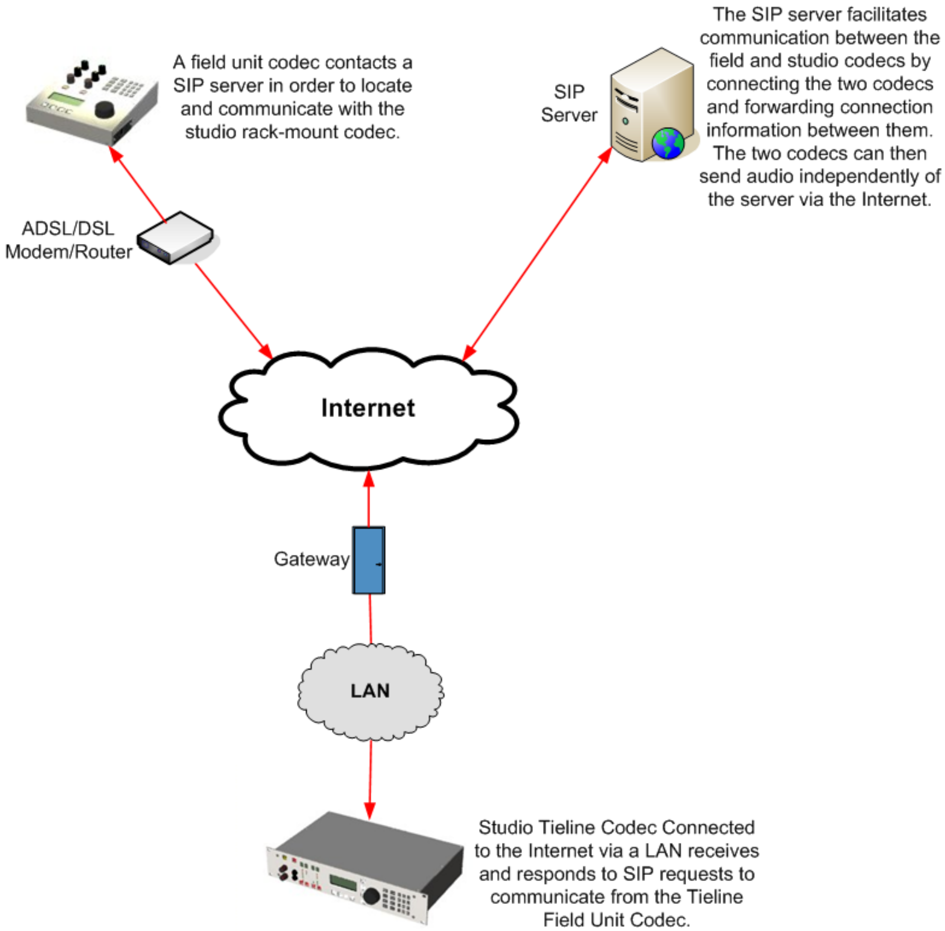
SIP only defines the way in which a communication session between devices should be managed. It does not define the type of communication session that is established.

As an example, broadcast audio codecs can use SIP to make peer-to-peer connections between two codecs. In this scenario, IP addresses are used to dial between two codecs and then SIP uses Session Description Protocol (SDP) to negotiate the features to be used during a call. This may include the bit-rate of the connection and the algorithm to be used.

One of the challenges in broadcasting audio over the Internet is creating reliable connections that support remote broadcasting from different locations. In the past, broadcasters have used IP addresses as a method of connection but this has been complicated for two reasons. IP addresses that are dynamically assigned often change and are therefore not reliable for dialing. In addition, private IP addresses, such as those assigned over private LANs, cannot be dialed directly by a device outside a private LAN. This principle is very similar to dialing a phone's extension number within a PABX system. Extension numbers within these systems cannot be dialed directly from outside the PABX.

To solve this requires either the programming of a static public IP address into a device like a codec, or the use of Network Address Translation (NAT) and port forwarding to avoid firewall connection issues.

SIP can be used to work around this problem by using a SIP server (registrar) to act as a gateway between public and private IP networks. The first step is to connect each SIP-compliant device to a SIP registrar which assigns SIP addresses (similar to an email address) to a device. Once two devices are SIP-registered they can find each other by connecting to SIP servers and exchanging connection information. This process is displayed in the following diagram.



When a SIP call is initiated, a SIP server establishes a device's location, determines its availability and, negotiates the features to be used during a call. Audio is then streamed according to those parameters.

Using SIP in Broadcast Audio Applications

In many ways the future of SIP technology development for broadcasting is being driven by the different approaches required for audio contribution and remote outside broadcast applications.

Tieline Technology, a leading broadcast audio codec manufacturer, saw the potential of SIP several years ago and since 2005 has been working on wired and wireless IP solutions for broadcasters using SIP.

Tieline and its partners in the Audio-via-IP Experts Group see SIP as the future of IP connectivity in broadcasting. They recently worked together with the EBU as part of a project group to test and develop EBU-approved standards for audio contribution and broadcasting over IP. These standards apply to signaling and network profiles, audio coding and transport protocols.

The EBU project group developed an interoperability standard between manufacturers based on open IETF (Internet Engineering Task Force) standards, which included SIP as a signaling protocol.

Audio codecs that use SIP and are EBU compliant should all be able to connect to each other more easily than in the past. As a result, major broadcasters in Europe will only consider purchasing IP codecs that have implemented these EBU standards.

Recent interoperability testing by Tieline and eight other European codec manufacturers found that they were all able to connect over IP using SIP. These tests used peer-to-peer connections that don't use SIP servers.

On the other hand, SIP-compatible audio codecs registered to SIP servers can simply dial and connect to other codecs and VoIP devices without knowing their IP address. This hugely simplifies remote broadcasting over IP and is particularly useful when broadcast locations are constantly changing.

One current issue is that most current SIP servers are configured for VoIP traffic and provide low quality audio using G.711 or GSM algorithms. For broadcast codecs to take advantage of SIP server connectivity, the SIP server should support higher quality algorithms such as MPEG, AAC and other proprietary ones. This requires a broadcaster to use a more compatible SIP server or transversal server to negotiate higher quality connections.

Broadcasters will also face some other challenges with SIP, including how to deal with Telcos who block SIP traffic. This is done to stop SIP traffic competing with other phone network products and is accomplished by blocking port 5060, which SIP uses to communicate between devices.

Negotiating all of these challenges will not happen overnight, but SIP is already providing interoperability between different codec brands in a multitude of situations. The influence of SIP is certain to grow as broadcasters work with industry stakeholders to further develop SIP functionality in years to come.