

# APPLICATION NOTE

**Audio distribution for large  
international events using  
the Phoenix Studio  
Audiocodec.**

**Case study, EBU Sports  
Operations London 2012**



## AEQ PHOENIX AUDIOCODECS. APPLICATION NOTE 0-G

### Audio distribution for large international events using the Phoenix Studio Audiocodec. Case study, EBU Sports Operations London 2012

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#### 1. INTRODUCTION - OUR SCENARIO

Broadcast operations at large international events such as the Olympic Games always present challenges of scale. One of them is the distribution of thousands of audio circuits from different sources (commentary, International Sound, PA, etc.) to all of the Rights Holding Broadcasters. When using a common infrastructure for this purpose, it is necessary create a structure similar to the figure below and that we call "Commentary System".

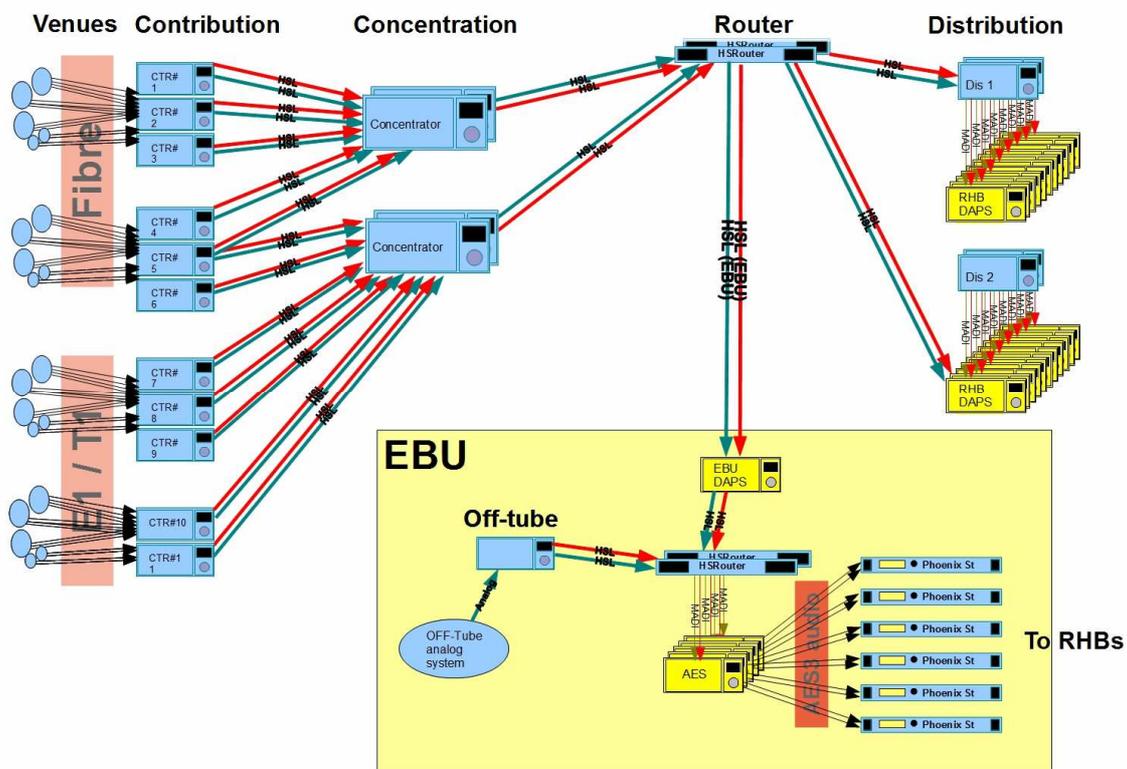


Figure 1: Commentary System Diagram

The audio signals are generated by the Host broadcasting organisation at the Rights Holding Broadcasters commentary positions at the different venues and through the AEQ commentary system. All circuits from every venue are sent to the IBC (International Broadcasting Centre), specifically named as CSC (Commentary Switching Centre) through AEQ BC 2000D routers. The transport of the circuits may be either a long distance (up to 160 Km) fibre connection or and E1/T1 multiplex. This is what is called Contribution.

Once all the circuits from the venues have been collected at Contribution they are be sorted and organised at Concentration. After that they go through an AEQ TITAN Router with the routing capacity of 5000x5000 audio circuits and that handles the circuit switching. The circuit switching ensures that each and every circuit reaches its respective destinations according to the Host Broadcasting Organisations planning. This switching is in most cases fixed, and following a long and very rigorous planning, but can also be dynamic with circuits shared by several users at different times.

Obviously the whole system is redundant, from the circuits origin at the venues up to the DAP (Digital Audio Panels - audio access points for each RHB in the IBC), including the main matrix. In the event of a main circuit path failure, the circuits will be automatically switched without signal loss to the secondary or backup path. The different audio and data boards that allocated in each system frame are also either duplicated or have backup boards that ensures the continuity of all circuits in the event of failure.

As indicated to the right in figure 1 above, all the individual RHBs or operators have their own DAP. However, in our case (EBU, European Broadcasting Union), it works a little bit different: the EBU switching centre at the IBC receives a high-capacity direct link from the main router in the Host Broadcaster Commentary Switching Centre. EBU distributes the audio circuits through their own AEQ TITAN matrix via other audio interface equipment to each associate or member broadcaster. These audio circuits are sent through dedicated equipment to the countries of destination using several different link types, like E1, IP or ISDN.



Figure 2: EBU switching centre, showing the AEQ TITAN frames (right), Interface Frames (center) and 24 Phoenix Studio for international circuits distribution, ready to be installed in London.

The E1 circuit distribution (not shown in fig 4) is accomplished directly with AEQ BC2000 frames and multiplexer boards, with the corresponding equipment at the associate or member broadcaster's facilities in their home country.

However, to transmit or distribute the audio circuits by IP and ISDN, it is necessary to use an audio-codec. This equipment should comply with the most demanding requirements for high quality, low delay audio and reliability of transmission. It is also required that the equipment automatically recovers and re-starts the communication in the event of a temporary transmission line failure. Further, the optimal channel usage is very important. The AEQ Phoenix Studio audio-codec is ideal for this application as the reader will become aware.



Figure 3: AEQ Phoenix Studio Dual Channel Audiocodec

In addition to circuit multiplexing and transmission, EBU provides many other services to its associate or member broadcasters. Among others, circuit control and monitoring, possibility to insert coordination messages into the circuits with special tools, etc. There are also special commentator positions that are not located at the venues but at the EBU IBC offices. The audio circuits are connected directly to the EBU matrix. These commentary positions are called “off-tube” positions.

## 2. AUDIO TRANSMISSION TO THE EBU ASSOCIATE OR MEMBER BROADCASTERS

Depending on the resources, the infrastructure and the number of commentators that each broadcaster has, the number of circuits to transport from IBC varies, as well as the telecoms infrastructure used for transmission. At the London Olympic Games, EBU will primarily use IP, but a handful of ISDN links will also be required. Both types of links are peer-to-peer (from the Audiocodec A to another A', B to B', etc.) The circuit transmission is accomplished using a high quality, low latency encoding algorithm – very much improved compared to the traditional circuits using G722 (although this algorithm is still very much used). For the occasion EBU is using the AEQ-LD+ algorithm developed by AEQ, with an audio bandwidth greater than 15KHz and with low latency.

As for the audio inputs and outputs, we decided to use the AES3 digital interfaces in order to provide an end-to-end digital transmission. This way, no further A/D conversions are required from the moment of the A/D conversion of the commentators microphone inputs up to the D/A conversion at the broadcasters mixing console or, ultimately, at the TV viewer's TV Receiver, maximizing audio quality.

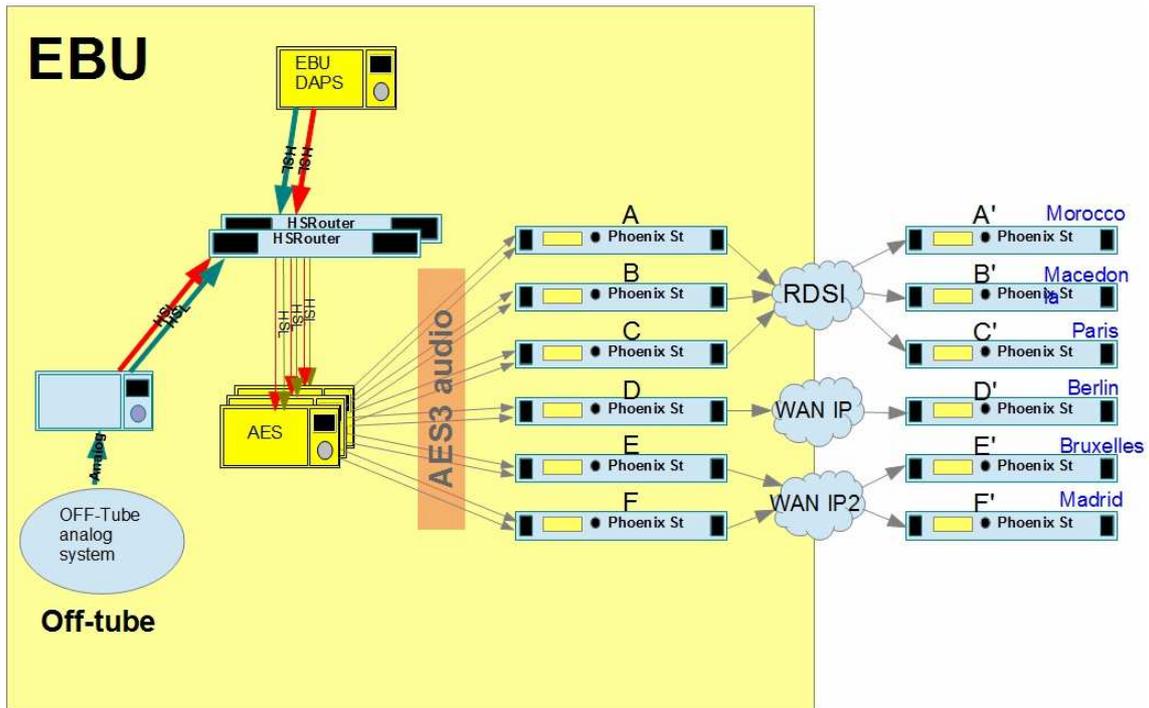


Figure 4: EBU Specific system diagram at IBC, including circuit distribution for the associate or member broadcasters

## 2.1. Sending IP Circuits

Broadcasters with greater resources use an IP infrastructure for their broadcast circuits. This consists in a high-availability, high-quality WAN network with layer 2 quality of service. The audio-codecs in London and the remote ones (located in the broadcasters home countries) are all configured to operate within the same logical network – all units have an IP address in the same range.

The table below shows the broadcasters that are using IP transmission for their Audio circuits, indicating the number of circuits and the number of Phoenix Studio audio codecs used in their home countries.

<p><b>AZICTI (Azerbaijan):</b> 1 x 15 KHz circuit, 1 x Phx. Studio</p> 	<p><b>BABHRT (Bosnia - Herzegovina):</b> 3 x 15 KHz circuits, 2 x Phx. Studio</p> 	<p><b>EGRTU (Egypt):</b> 2 x 15 KHz circuits, 1 x Phx. Studio</p> 
<p><b>TFRSRF (France):</b> 2 x 3.5 KHz circuits, 2 x 15 KHz circuits, 2 x Phx. Studio</p> 	<p><b>GRERT (Greece):</b> 1 x 15 KHz circuit, 1 x Phx. Studio</p> 	<p><b>BGBNT (Bulgaria):</b> 1 x 15 KHz circuit, 1 x Phx. Studio</p> 
<p><b>LVLTV (Latvia):</b> 3 x 15 KHz circuits, 2 x Phx. Studio</p> 	<p><b>MERTCG (Montenegro):</b> 2 x 15 KHz circuits, 1 x Phx. Studio</p> 	<p><b>BYBTIC (Belarus):</b> 2 x 15 KHz circuit, 1 x Phx. Studio</p> 
<p><b>DKTV2 (Denmark):</b> 3 x 15 KHz circuit, 2 x Phx. Studio</p> 	<p><b>DZENTV (Argelia):</b> 1 x 15 KHz circuit, 1 x Phx. Studio</p> 	<p><b>ESRNE (Spain):</b> 4 x 15 KHz circuit, 2 x Phx. Studio</p> 
<p><b>HRHRT (Croatia):</b> 2 x 15 KHz circuit, 1 x Phx. Studio</p> 	<p><b>ISRUV (Iceland):</b> 1 x 15 KHz circuit, 1 x Phx. Studio</p> 	<p><b>LTLT (Lithuania):</b> 2 x 15 KHz circuit, 1 x Phx. Studio</p> 
<p><b>MASNRT (Morocco):</b> 2 x 15 KHz, 1 x 7.5 KHz circuit, 2 x Phx. Studio (ISDN)</p> 	<p><b>MKRTV (Macedonia):</b> 2 x 7.5 KHz circuit, 1 x Phx. Studio (ISDN)</p> 	

Since the Phoenix Studio provides two transmission channels, it is possible to transmit two different programs using one piece of equipment per peer and even using a single Ethernet interface. This means that each unit is only occupying one port on the network switches, which becomes very interesting at the London side of operations where available ports on switches are scarce due to the concentration of equipment. Further, it really saves on cabling when the number of equipment sharing the same switch is optimized.

When using a single Ethernet interface, both audio channels will share the same IP address and it is necessary to distinguish the audio channels in some way. In order to do so, different RTP ports are configured, i.e., the “A” equipment, in London, which has IP address 172.26.33.55 will send two audio channels to the “A” equipment in Paris, with IP address 172.26.33.34. Channel one from “A” equipment will send audio through port “5004” (the same in the opposite way), and channel 2 will run over port “5008”. Note that the channels are full-duplex and with the same quality in both directions.

The signalling method chosen is SIP (without using Proxy). As opposed to a system using only the RTP (Real Time Transfer Protocol) protocol, when applying the SIP (Session Initiation Protocol) protocol there is no need to go through the process of establishing a connection at both ends simultaneously. One of the codes (usually the master, located in London) can make “calls” to the remote unit and that one can be configured to answer automatically, avoiding the need to have operators on both sides of the network to establish the connection. Once the connection has been established, the audio streaming is accomplished using RTP and RTCP (Real Time Control Protocol) protocols.

Each equipment will have an identifier or “URI”, e.g. <equipment\_name>@<ip> = [Phoenix\\_233@172.26.33.55](mailto:Phoenix_233@172.26.33.55). These URI’s can be stored in the Phoenix Studio agenda with a more indicative name (e.g. “Audiocodec Paris 1”), in order to simplify the process of calling.

Additionally, the fact that Phoenix Studio has encoding algorithms like the AEQ-LD+ including stereo modes, that can operate independently for the left and right channel encoding and de-coding, duplicates the equipment effective capacity for this particular application: since connection will be accomplished peer-to-peer it's possible to establish up to 4 independent bi-directional circuits with only 2 units. It should be noted that this would be impossible to accomplish in, for example, MPEG LIII Stereo, since this encoding algorithm typically use cross-channel information to reduce the required bandwidth and would therefore not be suitable for sending independent programs (not stereo), and will produce crosstalk and other harmful effects to the audio.

The encoding algorithm selected to be used by the broadcasters using IP is the high-quality family AEQ-LD+ 15 KHz audio bandwidth, fs 48 KHz @ 192Kbps bit-rate (384Kbps in stereo mode).

The Broadcasters that are using the traditional ISDN BRI's are in some cases also using the AEQ LD algorithms for 15 KHz circuits and the G.722 modes that provides 7.5 KHz audio bandwidths, but are limited to a maximum of two audio circuits per Phoenix Studio pair in peer-to-peer configuration.

The AEQ Phoenix Studio audio-codec is also equipped with adaptive buffers in reception that minimizes the jitter effects (variable delay between packets) and datagram loss in the IP network. This saves the users from having to configure too long fixed buffers that in most cases only generate unnecessary constant delay in the transmission.

## 2.2. Sending ISDN Circuits

The Phoenix Studio audio-codec is a multi-interface system, and also allows establishing connection over for example ISDN networks, using optional telecoms interfaces.

Some Radio/TV carriers, either limited by availability or costs of telecoms infrastructures in their countries or due to previous experience, prefer to use ISDN lines (Integrated Services Digital Network) for their program audio circuits.

In London the broadcasters that have chosen to use ISDN are:

<p><b>MASNRT (Morocco):</b> 2 x 15 KHz, 1 x 7.5 KHz circuit, 2 x Phx. Studio (ISDN)</p> 	<p><b>MKRTV (Macedonia)</b> 2 x 7.5 KHz circuit, 1 x Phx. Studio (ISDN)</p> 
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ISDN basic access has two “B” traffic channels, with a 64Kbps capacity each one, plus a C channel for internal signalling.

Phoenix Studio allows both B channels to be aggregated into one 128Kbps channel that enables high-quality audio transmission. To transmit two 128Kbps channels, it is therefore required to have two ISDN BRI's, but not two Phoenix Studio units, since each audio-codec has the capacity to allocate two optional telecommunication interfaces.

In this case, MASNRT, the national broadcaster of Morocco, is using 3 x ISDN BRI on two Phoenix Studio Audio-Codecs, allowing for 2 15KHz circuits plus one 7.5KHz circuit.

## 2.3. Failover protection

In a broadcast event of this nature, it is paramount that the system is fault tolerant and that it also provides features of automatic recovery.

Phoenix Studio offers several recovery methods: Backup, Auto-disconnect when detecting lack of data (RTP Inactivity Mode) and auto-redial. This makes the unit an ideal option for our application. Some of the recovery methods according to the failure type, are explained below.

### Backup:

Each channel can be configured to perform a backup reverting to a different telecoms interface in the event that the equipment detects an audio packet loss (when IP communication is used) or a remote hang-up or a call error during an established call (when ISDN communication is used). This backup can be real or not, that is, could really have a secondary transmission path or not. In any case, configuring it signifies an advantage because the equipment, in an error scenario, will retry the call for the main interface with the indicated intervals and number of retries.

If the unit is not capable to re-establish the communication it will run the backup interface, repeating the retry process configured as described above and then revert to try on the main interface again. This guarantees that the equipment continuously will try to return to the main interface when for example a transmission line fails.



Figure 5: Backup Configuration

When using a call-oriented protocol, such as ISDN or SIP, it is only necessary to configure the backup function for the audio-codec that is normally initiating the communication (in this case, the audio-codecs in London).

### Auto-disconnect due to lack of data packets (RTP Inactivity Mode):

It is important that upon an IP connection loss, which implies that the RTP packets do not reach their destination, the equipment will finalise the communication in order not to remain permanently busy. Therefore all the Phoenix Studio audio-codecs in IP peer-to-peer mode have been configured with this backup mode. It is recommended to limit the RTP inactivity time to maximum 15 seconds.



Figure 6: RTP Inactivity detection configuration

### Auto-redial:

Another useful Phoenix Studio feature in order to maintain communication at all cost is the start-up auto-redial. Activating this option, the result is that upon for example a power failure or an equipment reset, the audio-codec will try to re-establish the connection automatically upon start-up.

In the event of this occurring, the “caller” side (in our case London), will call the remote equipment upon booting, restoring the communication. It should be noted that for this backup mode to be entirely successful it is necessary that the RTP Inactivity Mode is activated for the remote peer, so that the previous call is released.



Figure 7: Automatic re-dial function configuration

In the event that the RTP Inactivity Mode is not activated on the remote or unattended peer and the auto-redial is not enabled, the communication will not be re-established. However, the backup function that is activated on the “caller” side (London) will re-call and the communication will be re-established.

Summarising, the configuration that we have adopted for the London operations is that the Backup transmission path is enabled (on V.35 for example) in London as well as the auto-redial (Redial on Interrupted Call). The remote peer does not have these options active. However, the RTP Inactivity Mode will be ON for all the peer-to-peer equipment used, i.e. both sides.

Everything explained above also applies to the ISDN links, although the RTP Inactivity Mode has no effect and would be the ISDN PBX or network that will disconnect the call due to line errors or a failing peer.

With this configuration is achieved a transmission link with very high possibilities of recovery without any intervention (and as long as transmission paths continue being available). The associate or member broadcaster will only suffer a temporary audio loss until the communication is re-established.



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