Practice and Reflections on the Construction of a Typical 800 m^2 Broadcast Studio Audio System: Postprint

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Abstract

With the implementation of broadcasting 4K/IPization at CCTV and some provincial stations, questions arise regarding the upgrade and renovation of audio systems for large and medium-sized studios in municipal television stations. In the current environment where baseband signals remain extensively utilized, this paper examines whether IP-based approaches should be considered and how to balance such considerations with budgetary constraints. By integrating the present situation and technical renovation requirements of the 800-square-meter studio at Taizhou Broadcasting and Television Station, we explore and implement a feasible solution for the current stage from perspectives of system design, equipment selection, and practical application, aiming to provide reference value for similar projects.

Full Text

Practice and Reflection on the Construction of a Typical 800 Square Meter Studio Audio System

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Abstract: As 4K/IP broadcasting begins implementation at CCTV and some provincial stations, questions arise regarding audio system upgrades for large and medium-sized studios at municipal-level television stations. With baseband signals still widely used, should IP methods be considered, and how can this be balanced with limited budgets? This paper examines the current status and technical renovation needs of Taizhou Broadcasting and Television Station' s 800 square meter studio, exploring and implementing a feasible solution from perspectives of system design, equipment selection, and practical application, hoping to provide reference for similar projects.

Keywords: studio; main/backup mixing console; sound reinforcement; distributed production; IP interface
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1. Project Background and Pre-Renovation Status

Taizhou Broadcasting and Television Station operates multiple production platforms including various studios, a converged media center, and OB vans. Among these, the 800 square meter studio is the largest, serving as a critical production platform for variety shows, award ceremonies, and large-scale cultural galas, handling both live broadcasts and recordings. The studio's video system utilizes a joint OB van setup for recording and broadcasting—a common configuration at municipal-level stations that saves on capital investment while improving equipment utilization. A relatively comprehensive audio system was also constructed.

The studio' s original audio system had been in service for over a decade. The sound reinforcement employed a stereo configuration with main speakers comprising two sets of Turbosound 12-inch two-way loudspeakers. While this design adequately met the needs of speech, vocal, and drama productions at the time, contemporary variety, music, and dance programs demand significantly more low-frequency extension.

The mixing console system consisted of a Soundcraft Si3 digital console as the primary system, capable of processing 64 channels, and an ALLEN&HEATH GL3300 analog console as the backup with 40-channel capacity. The backup system could only handle partial audio source redundancy and required manual source checking and patching during activation—a cumbersome process. Although generally stable, the aging console system began exhibiting intermittent noise from faders and knobs, with overall system noise floor gradually increasing. As production scales expanded, the increased usage of wireless microphones and digital peripheral devices revealed that the original system' s interface capacity and channel count could no longer meet program requirements. Additional limitations included insufficient flexibility for system output routing, lack of rapid emergency switching during live broadcasts, inability to accommodate different mixing requirements across multiple operator positions for the same program, and inadequate visual presentation of channel parameters (routing, bus assignments, EQ, panning, gain values, etc.) on the console surface—all of which failed to meet live broadcast demands.

2. Audio System Renovation Requirements

Based on these identified issues, we established the following requirements for the new system renovation:[1] The main mixing console needed to be a medium-to-large scale digital console with intuitive core parameter display, rapid operational response, and scalability, featuring processing capacity of no fewer than 96 channels (80 fully-processed input channels + 16 fully-processed bus channels). Considering the technical renovation cycle, the system needed to remain viable for eight years or more. The technology also required forward-looking design, particularly since IP system concepts have already been implemented at major stations like CCTV, necessitating IP connectivity capabilities for our system. The main/backup switching system needed to handle not only multi-channel broadcast systems (2-channel stereo + 5.1 surround sound, totaling 8 channels) but also sound reinforcement systems (2 main channels + 1 subwoofer + 4 stage monitors). The sound reinforcement system required a 2.1 configuration, with careful evaluation and selection between line arrays and point source systems. The setup needed to include left/right stereo main channels, independent subwoofer channels, plus stage monitors, rear stage fill, and lip fill speakers.

3.1 Sound Reinforcement System

The selection range for sound reinforcement loudspeakers is extensive, with numerous domestic and imported brands available. Our primary selection method involved inviting suppliers to provide products for actual listening comparisons.

Regarding the choice between domestic and imported brands, some domestic manufacturers currently offer respectable performance in terms of wired microphone response sensitivity and equalization adjustments, as well as music playback quality, dynamic range, layering, and clarity for large-scale symphonic material. For portable systems, we would readily choose domestic brands, which offer better portability considerations and significant price advantages while maintaining acceptable sound quality. However, for constructing a core studio requiring over a decade of service life, we favored mid-range products from major imported brands. As is well known, studio loudspeakers are permanently installed, and reinstallation involves substantial engineering work. After initial installation and calibration, the angles are fixed; any maintenance requiring disassembly and reassembly would necessitate readjusting angles and sound field calibration—a time-consuming and labor-intensive process that, without manufacturer technical support and measurement equipment, typically fails to restore the original performance level. In terms of factory warranty duration, major imported brands offer significantly better commitments than domestic brands; for example, d&b provides a 5-year factory warranty, demonstrating confidence in their material and manufacturing quality.

For the selection between line arrays and point source systems, we utilized EASE software to conduct sound field simulations across different frequencies, requiring overall uniformity within \$$\pm 3dBat16metersfromtheloudspeakersintheaudiencearea, achievingClassIaco$

seeoriginalpaper].*Costcomparisonsrevealedthatpointsourcesystemsweresubstantiallylessexpensivethanline*
pattern, whereas achieving a 50° vertical angle with line arrays would require at least 4-5 enclosures, further increasing costs. While more affordable dual 6/8-inch compact arrays could be selected, their frequency response and tonal character cannot match the natural integrity of 10/12-inch units' lower frequency limits. Consequently, we selected 12-inch two-way point source loudspeakers as the main system, employing two enclosures vertically arranged to cover near and far fields, driven by separate amplifier channels to enable independent frequency response and delay adjustments.

Considering subwoofer enclosure size, weight, and installation/maintenance convenience, we chose two single 18-inch units for low-frequency coverage. Stage monitor loudspeakers were selected as 12-inch coaxial models featuring high-frequency dispersion horns; we employed the same model for flown monitors to provide fill for the rear stage area.

Regarding amplifiers, the gap between domestic and imported brands remains evident, primarily in intelligent control capabilities. Major imported brands typically offer amplifiers and loudspeakers from the same manufacturer with built-in acoustic DSP parameters, enabling remote control and monitoring (including temperature, level, signal switching, DSP parameters, scenes, etc.), with some even integrating remote diagnostic and testing functions. While some domestic loudspeaker brands manufacture their own amplifiers using Class D technology, achieving increasingly comparable stability and sound quality to imported brands, they clearly lack many software functionalities, reflecting domestic shortcomings in digital acoustic R&D and application.

3.2 Mixing Console System

The mixing console serves as the core of the audio system, with most operations performed on its surface. A well-designed console significantly enhances system usability, flexibility, and rapid response capabilities for complex requirements.

We reviewed and compared first-tier broadcast consoles including STAGETEC, LAWO, TAMURA, and CALREC, while also referencing mainstream live sound consoles such as YAMAHA PM series, SSL LIVE series, SOUNDCRAFT, DIGICO, and ALLEN&HEATH. Broadcast consoles are relatively more expensive; although our application involves sound reinforcement, the studio primarily focuses on signal production, and most live sound consoles do not support 5.1 surround bus formats. Therefore, we leaned toward broadcast console options.

The LAWO MC236 console surface comprises three modules, each containing a 16-fader module and a 22-inch touchscreen, with modules operating independently—failure or reboot of any single module does not affect others. Additionally, it supports 2110-30/31 IP interfaces and 2022-7 stream switching, satisfying our IP reservation requirements. The console also includes numerous practical production features such as AUTOMIX (automatic mixing), UPMIX (stereo up-mixing to 5.1/7.1 formats), and AFV (audio-follow-video functionality). These user-friendly plugin features significantly favored LAWO in our selection process. Market research indicated LAWO consoles enjoy a solid reputation among broadcast consoles. While other consoles offered distinct advantages—for instance, YAMAHA enables automatic gain compensation between two consoles with automatic mixing plugins, and STAGETEC features more rotary knobs and high-dynamic-range microphone interfaces—comprehensive evaluation within budget constraints revealed that LAWO' s MC236, with 40 physical faders and 192-channel hardware processing capacity, better accommodated future IP system upgrades, networked applications, and interface expansion. This clearly demonstrated superior hardware and technological advantages, more closely aligning with our operational requirements.

The 800 square meter studio undertakes major events including large-scale variety shows and award ceremonies, with program live streaming becoming increasingly regular. System design must satisfy the complex and diverse demands of variety show recording while ensuring live broadcast safety. Accordingly, we designed a hybrid architecture combining primary/backup systems with a secondary distributed system, with the schematic diagram shown in Figure 2 [Figure 2: see original paper].

4.1 Signal Sources

Stage box connections in the performance area constitute crucial signal sources for variety studios. Wired microphone signals from bands or guests on stage are transmitted via multi-core cables from stage boxes to passive splitters in the control room, simultaneously distributed to both main and backup consoles.

Wireless microphones represent core audio signal sources, with over 20 channels in this system also fed through passive splitters to both consoles. We generally consider console microphone preamplifiers superior to wireless receiver outputs, so we configured receivers for microphone-level output to further enhance audio quality.

Audio playback peripherals primarily include computer and 360 hard disk player signals, which can be distributed via analog to the backup console and digital to the main console.

Video-derived signals mainly comprise audio from VTR playback and de-embedded external audio, also distributed digitally to the main console and analog to the backup console. Single-channel digital outputs (such as de-embedded signals) are distributed through digital passive splitters before being sent to both main and backup consoles.

4.2 Synchronization

Synchronization signals are derived from the Black Burst (BB) signal provided by the video system. The main console can accept BB sync input and convert it to WordClock to lock the backup console. In the future, the main console will also be able to receive Precision Time Protocol (PTP) clock from the network.

4.3 Network and IP Connectivity

The console features three RAVENNA interfaces on its rear panel. The first two ports are reserved for primary and backup switches for automatic 2022-7 switching; they can also connect directly to the video core switch. As 4K/IP systems represent future development trends, once the video system upgrades to IP, direct network cable connection to video IP devices will be possible. Currently, the third RAVENNA port connects to a COMPACT I/O interface box using network cable and fiber as mutual backup connections.[2]

4.4 Mixing Console Operation Modes

Since signal sources are distributed 1:1 to both main and backup consoles, they can operate completely independently and in parallel. During critical live broadcasts, the two consoles function in primary/backup relationship, with switchers selecting the A signal path—mixing is performed by the main console while the backup serves as a hot standby. For complex variety show recordings, the main and backup consoles can divide responsibilities: the main console handles production recording and broadcast signals while the backup manages live sound reinforcement signals, with both able to independently adjust microphone gains without mutual interference.[3]

4.5 Signal Output

Output signal paths include 8-channel PGM (main mix plus stem signals) for embedding or recording in the broadcast system, plus another 8-channel PGM feed for sound reinforcement (4 main, subwoofer, fill, and 2 monitors), all using digital signals through 8-channel switchers. An additional 8-channel switcher is configured for switching analog-format recording, monitoring, and IFB signals.

4.6 Spatial Layout

Equipment layout must consider both operational convenience and efficient physical space utilization. The control room is a narrow space converted from a catwalk, forcing equipment and main/backup consoles into a horizontal arrangement, as shown in Figure 3 [Figure 3: see original paper]. The equipment rack, housing wireless microphones and patchbays, is positioned first, followed by the main console. Monitoring is placed directly in front of the audio engineer, with a playback operator station for multimedia source devices located to the right of the main console position. Further right are the backup console and second playback station, with all positions offering direct sightlines to the stage for better situational awareness.

4.7 Sound Reinforcement

The sound reinforcement system employs a 2.1 configuration with one speaker group on each side. Each group comprises two 12S enclosures in a vertical array: the lower unit covers the near field while the upper covers the far field, driven by independent DSP channels and amplifiers to ensure separate acoustic parameter adjustment. Factory-designed rigging hardware minimizes inter-enclosure interference. The subwoofer is center-flown, which, though less efficient than ground stacking (compensated by increased level), maintains a clean stage front and reduces maintenance requirements. Four M12 loudspeakers are configured as stage monitors for flexible positioning; the M12 features a coaxial design with a dispersion horn, providing concentrated acoustic energy coverage within the effective area. The same model is also used for flown monitors at the rear stage area. Two lip-fill loudspeakers are added at the stage edge to anchor the sound image while better covering the front row—typically reserved for leaders and VIP guests—creating a more natural listening experience without the sensation of sound emanating from above. The main loudspeaker layout is shown in Figure 4 [Figure 4: see original paper].

5.1 Practical Function (Automix)

After system completion, testing, and a one-week trial operation, the first live broadcast arrived—the "Government Hotline · Mayor Online" program, shown in Figure 5 [Figure 5: see original paper]. The stage layout featured two rows of guest seating on each side in an open space requiring sound reinforcement for clear communication. The host and guests required a total of 24 microphones; the production team brought 16 channels of Shure ULXD handheld units for temporary integration to supplement insufficient microphone quantities and provide emergency backup, placing considerable pressure on the system and audio engineer.

Operationally, any of the numerous handheld microphones could be active at any time, necessitating careful feedback prevention. The flown stage monitors covered the microphone pickup areas directly, yet only these flown monitors could adequately cover the guest area to provide clear monitoring, making feedback likely between microphones and monitors. Since the control room is located on a second-level catwalk, the engineer would hear critical feedback later than the audience, imposing higher demands on audio operations.

We employed LAWO' s Automix function, grouping microphones into Automix sets, which served two primary purposes. First, inactive microphones were automatically muted, reducing microphone spill while increasing system gain. Second, the host' s microphone could be assigned priority to suppress overly loud guests or those interrupting the host. Comparative testing showed that with Automix enabled, feedback was significantly less likely and the sound was cleaner with no apparent noise floor. With Automix disabled, microphones tended toward feedback threshold conditions with substantial noise floor. This difference measured approximately 6-8dB on fader scales, equivalent to a similar increase in system gain.

5.2 Practical Function (Faderstar)

Typically, only one audio engineer is on duty. With so many microphones in use, we needed to minimize their workload. Therefore, we implemented the Faderstar function: the PLAY button on the 360 hard disk player can be set to trigger relay control, connected to the console' s GPO. In the console software, we configured the fader for the corresponding 360 signal channel to trigger the GPO, enabling playback simply by raising the fader when entrance or award music is needed. After repeated practice and adaptation, this proved highly practical.

5.3 Real-time Sound Pressure Monitoring

With the control room located on a catwalk, controlling comfortable volume levels for the venue is challenging. Therefore, we introduced a sound pressure monitoring software—the 10Eazy system—with a test microphone placed in the venue. After calibration, it monitors sound pressure in real time, intuitively displaying SPL values during production to assist the engineer in maintaining overall program volume. This is shown in Figure 6 [Figure 6: see original paper].

6. System Characteristics Summary

This audio system technical renovation solution incorporates the legacy Si3 console while selecting a more advanced IP-architecture console, enabling seamless integration with baseband connections while adapting to future IP system connectivity. This approach can maintain technological advancement for the next 5-8 years, offering several advantageous characteristics.

6.1 Safety Meets Live Broadcast Requirements

Since the MC236 and Si3 consoles have equal and independent access to signal sources, both can initiate local gain control over the same microphone without mutual interference. Outputs pass through physical two-select-one switcher channels, so when the main console experiences critical failures such as system crashes, the backup can fully assume program production responsibilities, satisfying the requirement for two independent systems essential for safe broadcasting.[4]

6.2 Enables Distributed Production

Variety show recording involves numerous uncertainties and complex changes, imposing extraordinary demands on audio/video systems. For example, award ceremonies with important dignitaries prioritize live sound reinforcement; music programs require multitrack recording for post-production editing; gaming/esports programs emphasize monitor signals for players and referees. Addressing these diverse future program needs demands not only powerful equipment functionality but also system flexibility. Post-renovation, the two consoles achieve division-of-labor redundancy. Whether multitrack or stem recording is required, the main LAWO MC236 console' s 144 buses fully satisfy requirements, while the backup Si3 can focus on live sound reinforcement needs. This allows both consoles to control source gains according to different priorities, meeting individual level and balance adjustments to achieve distributed production.[5]

6.3 High-quality Surround and Stereo Simultaneous Broadcasting

Some important program recordings may involve surround sound sources or require surround production. The main MC236 console features both 5.1 and stereo bus formats, enabling independent mixing of both types. Alternatively, system switchers can control main and backup consoles to separately produce surround and stereo signals through division of labor. This approach yields stereo signal quality far superior to that obtained through surround downmixing.

6.4 Reserved Future IP Upgrade Interfaces

As an increasing number of manufacturers transition to IP interfaces, future systems will undoubtedly be based on switch connections. Therefore, this system reserves interfaces for connecting to future primary and backup IP networks, dedicated to peripheral equipment and video system IP connectivity. The main console can provide up to 3$ $\times 128 channelstothenetworkviaitsIPinterfaceswhilesimultaneouslyreceivingupto 3\times$ audio channels, supporting 2022-7 automatic switching redundancy.

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