AEQ

HD VoIP ON AIR
Telephone
System

SYSTEL
MAX

Scalable Talk-Show and Telephone Multi-Conference System
for Radio, Television, and Business Environments

Operation Environment

Field of application of the product

SYSTEL MAX is a “call-in” system and multiconference capability that drastically reduces the costs for this type of communications. It significantly improves the audio quality, increases the flexibility and integration with already existing telephone systems at the station or company. The investment required is very small and will be amortized very rapidly through simple cost saving.

Business telephone systems are rapidly migrating to VoIP technology, integrating IP switchboards or virtual, allowing access to new alternative telecommunication service providers.

Telephony or call-in systems for broadcast applications have until now been an isolated island with important operational costs and stagnant technology. SYSTEL IP allows to connect the broadcast telephone system to the current corporate PBXs, based on IP, avoiding maintaining conventional lines exclusive for broadcast.

SYSTEL MAX allows for VoIP connection of 4-wire lines from intercom matrixes or audio consoles in order to establish multi-conference circuits or external coordination in radio or TV stations.

Further, in a business environment it allows for the interconnection of several meeting rooms as well as audio routing between building locations and for example simultaneous translation systems, even if these are remotely located.

Background

Multi-line telephone systems for the broadcasting industry have been available for more than 20 years. AEQ has continuously been offering innovative solutions and in line with the available technology:

In 1994 AEQ developed the Systel 3000 conference system with control for digital telephone hybrids on conventional telephone lines in console multiplex format.

In 2004 AEQ launched the Systel 6000, with a new architecture: High quality ISDN lines with AudioCodecs, POTS, leased lines and point to point-to-point IP Audio (RTP). The system incorporated a 4-wire digital matrix that allows console format multiplex and multi-conferencing of up to 40 different channels.

In 2014 AEQ offered the the third generation: the system continues being built around a digital router and we are using lines from IP telephony systems implementing a flexible and dynamic control. Further, the call-in-queue is controlled simultaneously with the multiplex functions for the comfort of all users.

In 2018, a new “Enhanced” version was presented, incorporating a new engine with IP Dante local connectivity, a control terminal based in a IP phone with touch screen, and a new software application for TV coordination.

Now in 2025, with over 30 years of accumulated experience, this system is introduced based on the powerful AEQ Solaris multicodec engine. It guarantees, on one hand, scalability of up to 128 lines distributed across as many studios as needed, and on the other hand, the highest reliability, thanks to the implementation of the SIP protocol and the hardware platform used—as well as broad interoperability with a wide variety of PBX systems and SIP trunking.

Four basic concepts regarding VoIP

IP Telephony or VoIP
Currently, the way to functionally enable communications between conventional telephony and the majority of private switchboards (PBX): if a call is generated with a conventional phone, the generated audio signal will be converted into a digital signal, compressed and encapsulated under internet protocol (IP) within a gateway. From the gateway the signal will be forwarded to the recipient’s phone within a computer network - WAN. The audio that reaches the conventional phone, has previously travelled through the network to the “gateway” where IP packets have been converted into audio for the earpiece. If the phone is an IP phone, the phone itself generates and receives IP packets. In this way telephone voice signals are converted into and treated as computer data and flows through Internet networks via switches, routers, ADSL lines etc...

IP PBX
At this point, it is not difficult to imagine that a telephone is a kind of computer with specialized software. SYSTEL MAX will interact perfectly with an array of IP PBX systems available, including the generic ones that are based upon Asterisk and the commonly used in business environment such as Cisco Call Manager, Alcatel OXE, Avaya IP Office, etc.

VoIP Providers
These area Internet-based service providers that are able to route calls through the network with access to traditional telephony in different cities and countries, allowing international rates adjust to a little more than the cost of a local call. The VoIP service providers (their services) are accessed through a trunking IP (Internet access: DSL, cable modem, fiber optic, WIMAX ...). Some offer virtual PBX service: connect all IP phones to an office trunking with a switch without
the need of a switchboard.

SIP
SIP is a signalling protocol for VoIP (Voice over IP) to route calls between locations and equipment. SIP is also available in the AudioCodecs that meet the N / ACIP EBU (European Broadcasting Union) and in many “Softphones” that allows establishing calls from computers, PDAs, etc. using the telecommunication companies’ data networks. SIP allows both partners to negotiate and establish audio codecs high quality calls (HD) if both phones support it.

Main Features

• SYSTEL MAX does not operate based on hybrids, but rather on a 4-wire digital matrix: All lines can be on-air simultaneously in a program without any loss of quality.

• Significant cost savings can be achieved by connecting the complete system to an internet telephony provider or as extensions of the existing IP PBX already in service within the organization.

• SYSTEL MAX works by sharing IP lines flexibly and dynamically, supporting as many studios as allowed by the license, with audio over IP connection using the AES67 or Dante protocol.

• The SYSTEL MAX control terminals are extremely powerful, flexible, cost-effective, and user-friendly. The following can be used:

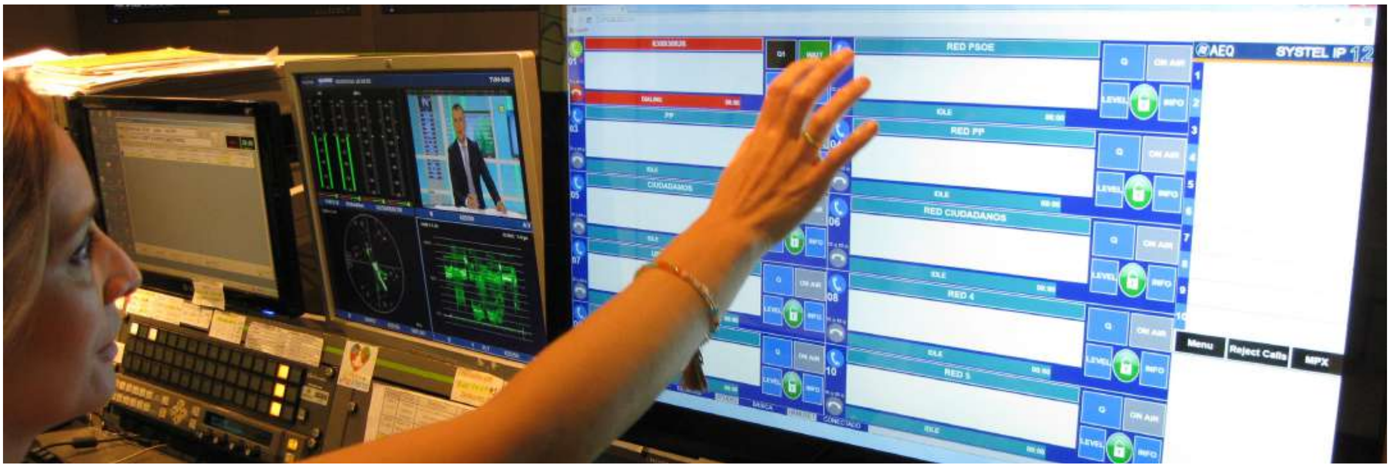
• An IP phone for calling and speaking with participants, along with an application that can be installed on any PC.

• A touchscreen IP phone for calling and speaking with participants, running a dedicated application.

• Several studios and work spaces can be defined. Multiple control terminals with internal and individual labelling and chat lines can be used in a studio, thus dividing the work among producers, technicians and talents.

• Possibility to set the number of audio signals arriving at the studio console, allowing for level adjustment either through this SW application or the fader of the mixing console.

• There are applications with different layouts and functions available in order to suit different types of operation.



Central element of the system

SOLARIS FOR SYSTEL MAX. The core of the system is a 19” rack-mounted unit:

SOLARIS is a multifunctional device which, when configured for SYSTEL MAX, can activate (according to the license) up to 128 IP phone lines or operator phone extensions, and 128 AES67 or Dante protocol IP inputs/outputs — enough for a multi-studio radio or television center, supporting dozens of studios depending on the configuration.

The unit functions as a set of multi-line IP phones with SIP protocol signaling. It is compatible with IP PBXs, SIP Trunking, and virtual PBXs. It also supports analog and ISDN lines through gateways.

In addition to the G.711 codec, it also includes G.722 wideband encoding (up to 7 kHz), which qualifies it as “HD”, and makes it compatible with N/ACIP audiocodecs and SIP phones (including all AEQ Phoenix codecs and most PC-based telephony software).

Control Applications

CONFIGURATION AND OPERATION SOFTWARE

SYSTEL MAX comes with a configuration application that creates the working environment (devices, lines, studios and programs) user groups and system operation. There are four different operation pieces of software in order to provide real-time control on the system:

SYSTEL IP ORIGINAL:

Based on call queues, suitable for radio production.

SYSTELSET+:

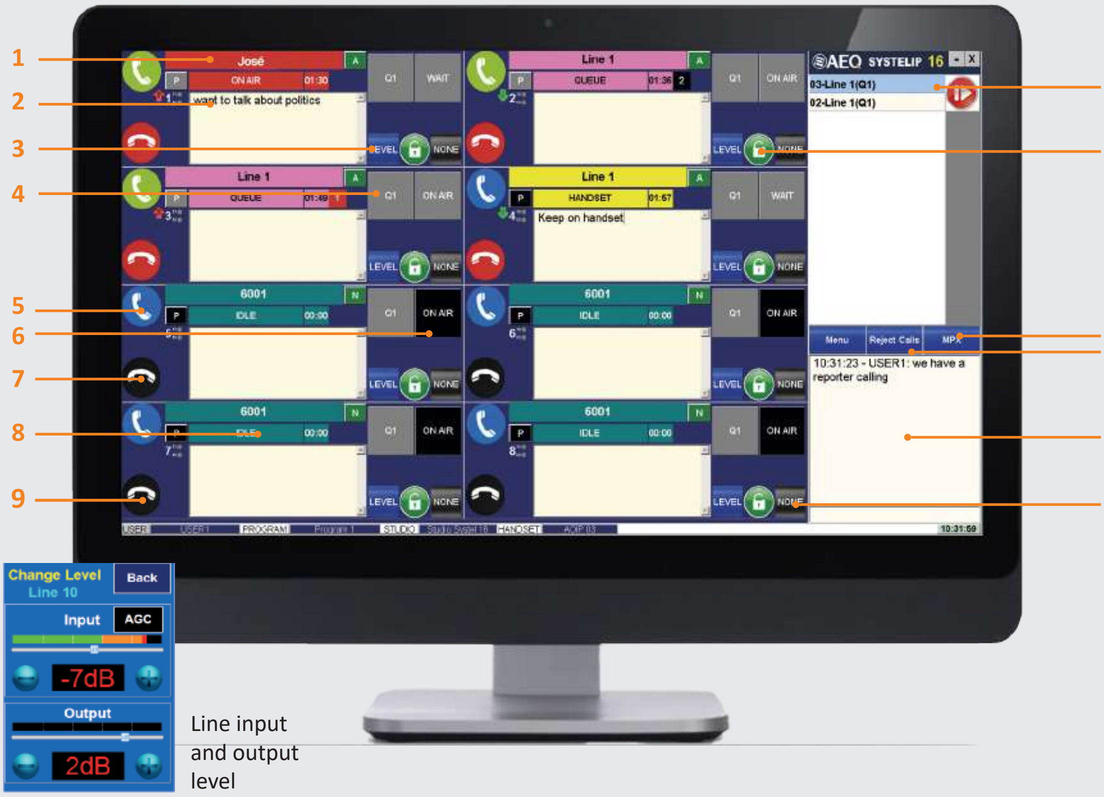
Integrated within the SYSTELSET+ divece, allows for a very flexible operation while avoiding the need for a PC in the control rooms or other reduced spaces.

SYSTEL IP TV:

With a similar appearance as the Original, it is based on multiplex. It is ideal to give access to Intercom systems or provide technical external intercom features, more commonly used in TV production.

SYSTEL IP ORIGINAL

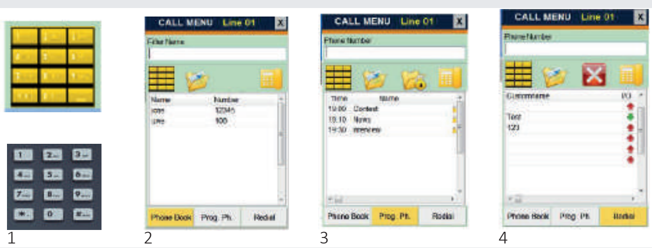
This is the best option in high-productivity environments where functionality is distributed in roles such as producers, controllers and talents. Calls are dialled or accepted, put on hold or pre-listened, their send and return levels can be adjusted, they can be diverted to auxiliary circuits, put on air, placed on hold, or hung up. The system can operate in call queue format, or alternatively several calls can be on air simultaneously in multiplex mode. All parties can exchange chat messages, tag lines and highlight annotations for each one of the calls, manage a contact list and call schedules. Lines can be shared among different programs and the layout is adapted to each program's available lines.



1 Editable remote party number and name.
2 To share info about the remote party.
3 Line input and output level adjustment.
4 Queue or fader selection and indicator.
5 Audio presence, Line number & call direction indicators.
6 Make, hang up calls and consult remote party.
7 Waiting or on-air call.
8 Hang up call.
9 Line and call status.
10 Status bar: User, Program, Studio, Handset and Clock.
11 A call queue is configured on every available fader. By clicking on the button, the next call on queue will be put on air. Calls can be re-sorted and consultations can be made from the queue.
12 Lock-protected calls are not removed from air when giving pass to another one.
13 All calls on air simultaneously.
14 The Reject Calls prevents incoming calls to be queued.
15 General chat between producers, controller and talents.
16 Information about the call that is on-air.

There are 4 possible ways to make a call:

1.- Dialing the number in the touch screen or the handset's physical keyboard.
2.- Through the program's call book.
3.- Using the programmed calls call book.
4.- Using the redial call book.



SYSTELSET+

This is the operation application embedded in the SYTELSET+ phone with touch screen. This provides for a very flexible operation and it is a great option to a PC in the control rooms or other areas with reduced space.

It is a valid option for all kinds of work environments, but its features are most appreciated in confined spaces (it only takes a surface of 26 x 17 cm, less than many other telephones), self-control rooms or in programs where there is not a large control staff, where calls are not the most important part of the program, but only one more important element.

On the terminal itself, using the function keys and the touch screen, calls are dialed or accepted, put on hold or pre-listened, their send and return levels are adjusted, they can be put on air, locked on air or hung up. The operational mode can be chosen between call queue or several calls on air simultaneously. The queue is built on the lines itself with an indication of next call to be put on air. It also allows for the management of a call book and call scheduling. Lines can be shared between different programs and the layout is adapted to the number of available lines at each moment, thus making the best use of the phone's touch screen.

SYSTELSET+, thanks to its reduced dimensions, allows for agile and flexible operation compatible with the Original Systel IP application in other terminals.

1 Remote party number and name.

2 Make, answer calls, consult remote party, line number, position in the queue, call direction, on-air. indicator.

3 Set this call first in the queue.

4 Menu.

5 Talk to next one.

6 Phone listening level.

7 Review call and put it on-air.

8 Hang up call.

9 Queue or fader selection.

10 Line and call status.

11 Time elapsed from last action on the line.

12 Use external micro -headphone.

13 Phone muting.

14 Put next call in the queue on-air.

15 Use hands-free.

MENU OPTIONS

By pressing the Menu key, lines display is compressed to the left, and the menu appears. This shows SYSTEL-SET+'s great flexibility and adaptation to usage habits.

SYSTELSET + Menu screen

General Menu options

Per program or studio:

Auto Answer: the system automatically answers incoming calls, leaving them on a preset queue or on hold, according to the configuration.

Block- All: Rejects all incoming calls.

Auto Conference: All calls are put ON AIR

Dump Mode: When a call is put on air, the former one us hung, without being put on hold(without removing the previous ones).

Page Lines: Using this option it is possible to talk to all correspondents at the same time, but they cannot hear each other.

Levels: Adjust each line's input and output levels.

Auto Next: If the ON AIR correspondent hangs up the call, the next one in the queue is put ON AIR.

Lock Show: locks the program so it cannot be closed by another user by mistake.

Particular Menu options

In each SystelSet:

Mute Ringer: mutes the outgoing phone ringer.

Pick-up Incoming: attends the oldest call on hold or ringing when the handset is picked up.

Auto Screen: sends the call to the queue instead of hanging it up when releasing the handset.

Direct Dial: automatically chooses a line for outgoing calls.

Direct Next: Calls are put on air one after another even if they have not been revised.

Advanced Settings

The following options are accessed when clicking here:

USA Mode: Configures labels and control buttons with typical USA naming.

Vertical Lines: Sorts the lines consecutively or even/odd.

Change Studio: Change studio.

Close Show: Close the program.

Logout: Leaves the application.

Control Applications

SYSTEL IP TV

Eases flexible and generic control of Systel MAX equipment, in what relates to external routing of intercom systems and other commonly used applications in TV production centers and similar environments. These are some of the specific features related to TV production added to SYSTEL IP Original ones:

- Accept incoming calls manually or automatically, label calls, put them on air or leave them in a multi-conference group.
- Leave calls listening to their assigned (N-1) feedback.
- Put the calls on air, routing them to the assigned internal audio circuit.
- Leave calls in different multi-conference groups, where all group members can participate talking and listening at the same time.
- Allow the operator to talk to all lines separately or, alternatively to all the group members at the same time.

1 Editable remote party number and name.
2 Line input and output level adjustment.
3 (N-1) feedback assignment.
4 Audio presence, Line number & call direction indicators.
5 Make, hang up calls and consult remote party.
6 Hang up call.
7 Call and line status and time elapsed for that communication.
8 Status bar: User, Program, Studio, Handset and Clock.

9 Reject Calls prevents incoming calls to be queued.
10 Activates the auto-answering function for incoming calls.
11 A 4-wire circuit is configured for each line, that can be assigned to faders or intercom ports.
12 Leave calls in different muti-conference groups, where all group members can participate talking and listening at the same time.
13 Activates and deactivates the permanent-call mode.
14 Handset input and output level configuration.

FRONTAL PANEL

SOLARIS, the Systel MAX engine, features only an information display on the front panel and a button to access additional screens with detailed information

1- Information display // 2- ON / OFF / standby switch and screen change // 3- Maintenance USB

BACK PANEL

1- 5 network ports (LAN1...LAN5) // 2- Maintenance USB // 3- Pre-listening output // 4- 2 AC power connectors with individual switches

FRONT INFORMATION DISPLAY

The unit is configured and operated through software applications and web pages. When powered on and in standby mode, an information display is shown. By briefly pressing the button to the right of the screen, you can access additional screens.

Standby screen
Top: Active IP interfaces and their speed.
Second row: CPU alarms (memory and temperature), power supply status, and time.
Bottom: Number of IP channels in use and number of available channels.

Firmware versions screen

5 Network interface configuration screens

Diagnostics screen

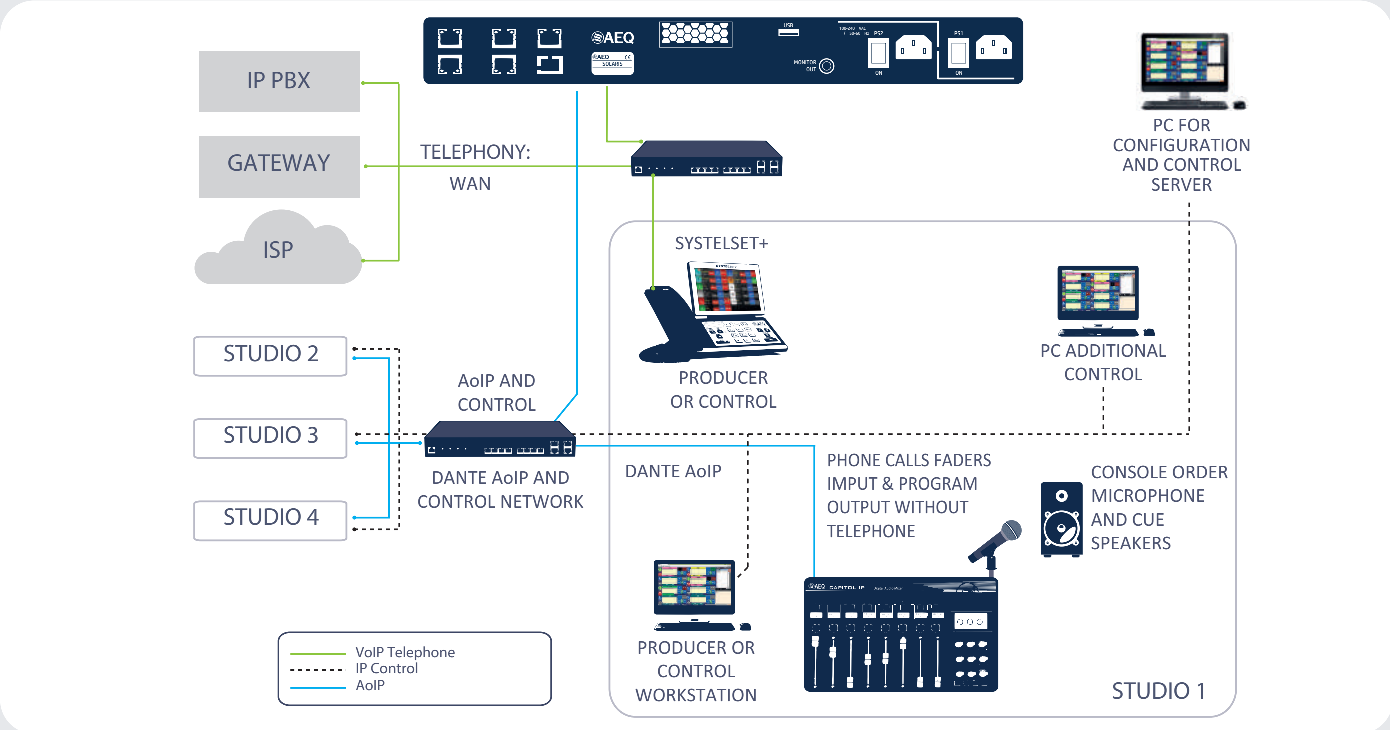
Radio Usage Scenarios

SYSTEL MAX and SYSTELSET+ control phone terminals “for four” studios

SYSTEL MAX functions as a set of IP telephones that can be shared flexibly and dynamically among four studios. Calls arrive via one ore more WAN Ethernet connections. Control commands are sent from the configuration PC and control server via the control Ethernet connection. Audio can be connected locally via the Dante network, which can be wired redundantly to two interfaces on the unit.

The control terminals access both the control server and the SYSTEL MAX device via IP. SYSTEL MAX allows the use of multiple IP telephone control terminals, for example, one or two per studio. The control technician can manage calls using talkback (orders) and CUE listening, especially when the SYSTELSET+ control terminals are assigned to producers. From each studio console, an auxiliary bus without telephone audio is sent, which is then combined with the rest of the telephone lines to provide individual return feeds to each phone.

Additional control PCs can also be installed.



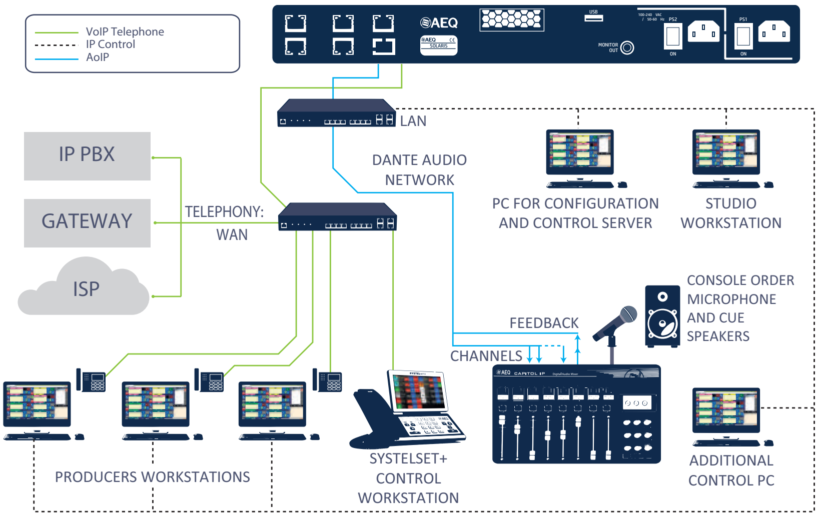
SYSTEL IP FLEX in a Multi-Studio Setup with Producers and Multiple Audio Feeds to the Console

The IP connectivity of up to 128 incoming and outgoing audio channels allows differentiated audio to be sent
to each studio using 1, 2, 4, 6 or more faders per console.

From each console, an auxiliary bus without telephone audio is sent to be mixed with the other phone lines, providing a dedicated return feed to each phone.

A total of up to 128 external IP telephone lines, plus internal lines for operators and producers, are available. Producers use standard IP phones and PCs to label calls and chat with the studio/host.

The control technician uses a SYSTELSET+ terminal, and if they need to follow the chat, the control application can also be installed on one of the control room PCs.



SYSTEL MAX, typical use of the Systel IP TV application in relation to an intercom system for outside broadcast coordination in a TV center.

TV broadcast centers are organized by studios. We are illustrating a center with four studios. Each studio has a sound console and several user intercom panels.

For control, we will add to the studio a PC with the Systel IP TV application and an IP phone to make calls and coordinate the studio's communications.

Shared among the four studios, we have an intercom matrix, a Systel IP FLEX unit, and optionally a PBX and the station's telephones.

There are two IP networks: the internal one (LAN) and the external one. A primary LAN port of the Systel MAX is connected to the LAN switch. The intercom matrix, the audio consoles, the user intercom panels, and the control PCs are all connected to the same switch. This simplified cabling, with a single internal network, can be split—if desired—into three separate networks: one for control, one for primary AES67/Dante AoIP, and another for secondary AoIP, since the device has five network ports.

To the WAN switch, another LAN port of the Systel IP FLEX is connected, along with the IP phones and the trunk or access to the telephone provider (ISP). If desired, the station's PBX can also be connected to the WAN switch. This is especially useful when we want to transfer calls from office phones.
Bidirectional audio channels are established between each Systel IP FLEX and the intercom matrix, up to a maximum of 128 depending on the license. The matrix will establish routes with the user panels and the audio consoles, as configured in the intercom applications Crossmapper and Live Crossmapper, creating workgroups or Party Line or more complex configurations if necessary.

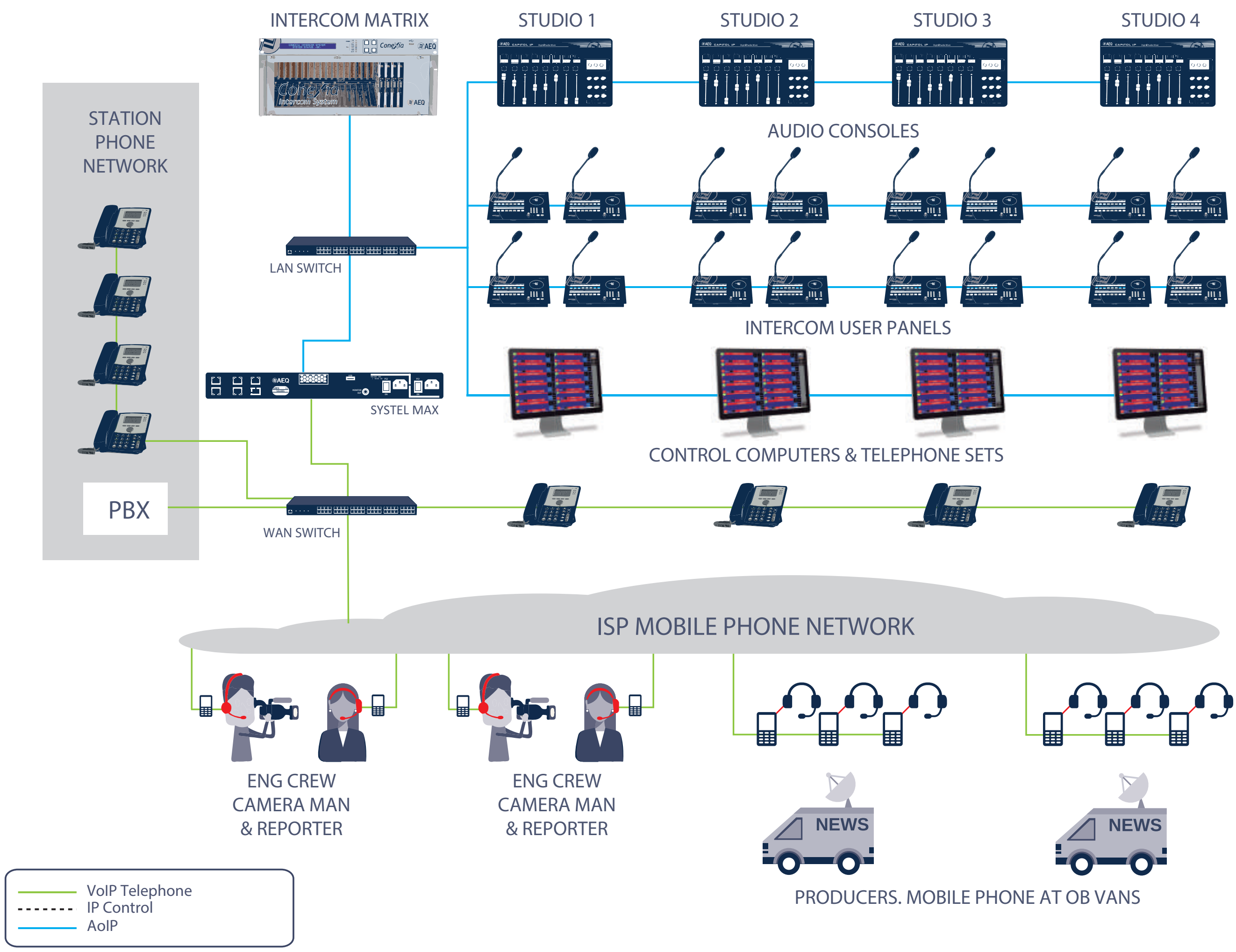
The Systel MAX lines, up to a maximum of 128 depending on the license, can be flexibly assigned to the studios, and calls can be initiated and received from the Systel IP TV application. The associated phone can be used to dial and speak before establishing communication. External calls can come in, and if auto-answer is enabled, the call enters the matrix and is routed to the assigned panels. Pre-programmed calls assigned to specific keys on the various user panels can also be set up.

The intercom matrix establishes bidirectional routes with the console and complex partyline-based routes with different panels, depending on the needs of each production.

The main application is coordination with ENG teams via mobile phones. The reporter receives program return and orders in their earpiece. The camera operator communicates using 4-wire through a headset connected to their phone, with a partyline group preconfigured in the matrix.

When working with an OB unit, at least three 4-wire circuits are established with the mobile unit, integrating them into a specific intercom group.

Systel IP FLEX also enables coordination between mobile devices, allowing the selection of who talks to whom at any given moment. It is also possible to decide who receives auxiliary signals, such as program return, for example.



System Components

Exclusive Components

"Engine" sized by license: SYSTEL MAX

All processing power and necessary connectivity are concentrated in a single rack unit "frame." It includes configuration and user software for an unlimited number of terminals.

Dimensioning and Licensing

- Internal Audio-over-IP (AoIP) lines in AES67 / Dante format are licensed in blocks of 16, up to a maximum of 128.
- External VoIP lines are licensed in blocks of 8.
- Internal VoIP lines for coordination and call-handling terminals—whether SYSTELSET+ or standard IP phones—are licensed in blocks of 4.
- The maximum number of VoIP lines is 128, including both external and internal lines.

The device's local audio connectivity is entirely based on IP networks:

- Audio circuits for AES67 / Dante consoles are established through the IP inputs and outputs of the console.
- To connect with consoles and other equipment that do not support AES67 or Dante inputs and outputs, converters such as AEQ Netbox or Dante AVIO must be used.
- To connect with internal PC audio, the DVS (Dante Virtual Soundcard) application can be used.

SYSTELSET+ Handset

Control terminal based on a Touch Screen IP phone running a configurable control application that can be adapted to the most diverse operating workflows. This eliminates the need for a PC at each workstation.

ANCILLARY COMPONENTS

To set up a Systel MAX system in a specific environment, it may be necessary to incorporate some audio, IT, or telephony accessories that are available on the market or even already in service at any radio or TV station or office. The requirements are not very strict, but if desired, AEQ can recommend or provide these approved accessories through our Systems Engineering Department.

Audio Interfaces

If the intercom or station audio matrix, or the studio's audio console does not have AES67 or Dante connectivity, Dante interfaces must be used. There are two types:

- Single- or two-channel. Powered via PoE and usually used off-rack. For example, Dante AVIO.
- Multichannel. Half or one rack unit. For example, AEQ Netbox.

Ethernet Switch

The system connects to one or more networks for control and AoIP, and another for VoIP. If these networks are not already in place for another purpose, an Ethernet switch must be installed for each network. In small installations, the networks can be unified.

PCs for Control Application, Configuration, and Databases

Almost any PC running Windows 10 or higher is suitable for installing the configuration and control applications. For the control application, a touchscreen is highly recommended. All-in-One PCs are especially suitable. For databases, any PC can be used, or a shared or dedicated server depending on the size of the installation. In small setups, everything can be installed on the same PC.

Operator Headset

Some producers who are constantly handling calls need the ability to use operator headsets. Both wired and wireless models are available on the market. They must have an RJ9 connector, so they can be plugged in in place of the SYSTELSET+ handset, which should be disconnected first. Some wireless operator headsets include an auxiliary port that allows the handset to remain connected simultaneously.

POTS FXO Gateway

Converts analog telephone lines into IP lines. Various models are available.

ISDN Gateway

Converts ISDN telephone lines into IP lines. Models are available for 1, 2, and 4 basic rate interfaces
(BRI), or for one primary rate interface (PRI).

IP PBX (Telephone Switchboard)

SYSTEL MAX does not strictly require a PBX, but it must receive IP calls from somewhere: a gateway, SIP trunking, or an IP PBX. Therefore, if the installation is being used as an opportunity to upgrade the station's telephony system and switch entirely to IP, we can recommend an IP PBX tailored to your needs.

IP Handset

SYSTEL MAX allows the use of compatible IP phones as a handset in place of SYSTELSET+. This has the advantage of allowing you to dial directly from the phone's keypad instead of from the application. Using the SYSTELSET+ as an IP handset offers the added advantage of having the control application integrated.

IP Phones

SYSTEL MAX doesn't need phone sets as it incorporates its own specific phone service terminals and can even use a PC microphone / speakers, or even the coordination circuit of an audio mixing console. But if you take the opportunity to migrate your station's telephony to IP, you should use IP phones in reception and all offices.

SYSTEL IP MAX

GENERAL FEATURES
SIP communication protocol: Compatible with VoIP trunkings, free PBXs like Asterisk and proprietary systems from major brands, SIP Phones such as Phoenix Pocket, Phoenix Lite, “N/ACIP compliant” audiocodecs like Phoenix Mercury, Phoenix Studio, Phoenix Venus, Phoenix ALIO, and FXO for POTS, ISDN, E1, and T1 lines.

Based on a non-blocking digital matrix, all lines configured in a studio can be live simultaneously in a program without any loss of quality.

Audio specications

- Dante inputs and outputs compatible with AES67. Dual IP LAN interface compatible with Dante redundancy. Synchronization is transported over the network.
- Telephone audio in G.711, 50Hz – 3kHz.
- HD audio with G.722 algorithm, 50Hz – 7kHz.

Configuration software and control server
Windows 10 and 11 OS.

Functionality (configurable for each user group)

- Assigns audio circuits, handsets, IP phones, and chats to the various studios uniquely.
- Renames circuits. Defines and manages contact lists, allows users to share, edit, and copy them.
- Defines PFL signals assigned to each studio.
- Defines auxiliary and master signals assigned to each studio.
- Configures initial audio levels for each line and each studio.
- Configures the format of client screens, defining the number of lines per program, operation in console mode, or with one or several call queues.
- SIP configuration for communication with IP-PBX switchboard, FXO gateway, external IP telephony provider (via Internet), or internal (on a LAN or WAN).
- Differentiates and protects functionalities based on user rights by activity type.

SYSTEL IP Original and SYSTEL IP TV control clients
Windows 10 and 11 OS.
Functionality (configurable for each user group).

- Establish calls by dialing numbers, SIP identifiers, or using contact lists and scheduled calls.
- Establish calls by dialing numbers on IP phones with IP handset functionality. Emit a visual and audible RING signal.
- Display the caller ID or number. Identify callers by their name in the contact list or add a temporary name.
- Manually or automatically answer incoming calls.
- Define and manage telephone directories, both general and private by program.
- Save new contacts in the directory.
- Create and manage scheduled telephone calls.
- Speak via headset or earphones with the person on the other end of the line.
- Put calls on hold while they listen to the program.
- Put calls on air so they can contribute to the program.
- Queue up on one or more faders the calls ready to go on air, allowing dynamic reordering and review.
- Assign a VIP attribute to a call to keep it on a dedicated fader.
- Change the headphone listening levels and the input and return levels of each telephone line in the studio.
- View the status of each telephone line, the duration of its current state, and where it is being routed.
- Tag calls. Chat between different controllers assigned to a program (Original version only). Manage blacklists. Block incoming calls.

SYSTELSET+ control client
Android OS.

Functionality (configurable for each user group).

- Establish calls by dialing numbers, SIP identifiers, or using contact lists and scheduled calls.
- Establish calls by dialing numbers on the phone.
- Emit a visual and audible RING signal.
- Display the caller ID or number. Identify callers by their name in the contact list or by a temporary name assigned from the SYSTEL IP Original application.
- Manually or automatically answer incoming calls.
- Save new contacts in the directory.
- Manage telephone directories, both general and private by program.
- Manage scheduled telephone calls.
- Speak using headset, speakerphone, or earphones with the person on the other end of the line.
- Put calls on hold while they listen to the program.
- Put calls on air so they can contribute to the program.
- Queue up on one or more faders the calls ready to go on air, allowing dynamic reordering and review.
- Assign a VIP attribute to a call to keep it on a dedicated fader.
- Change the phone’s listening levels and the input and return levels of each telephone line in the studio.
- View the status of each telephone line, the duration of its current state, and where it is being routed.
- Distinguish and protect, through user rights, the functionalities of producer, operator and presenter.
- Manage blacklists. Block incoming calls.
- Work with terminology and functions typical of American and European operating modes.
- Activate Dump Mode to hang up or not after being on air.
- Activate Page Lines to send alerts to all lines and receive simultaneous responses.
- Activate Auto Next to move to the next call when another is hung up.
- Activate PickUp Incoming to automatically connect with the oldest call when lifting the handset.
- Activate AutoQueue to automatically queue the call when the handset is hung up.
- Activate Direct Dial to skip line selection when making a call.
- Activate Direct Next to put calls on air even without answering them beforehand.

SOLARIS “Engine” SYSTEL IP MAX

5 configurable IP ports (LAN 1 to LAN 5) for external VoIP lines, VoIP lines for control phones, control network, primary and secondary AoIP Dante / AES67 network. RJ45 10/100/1000 Mbps.

General features

- Diagnosis via SNMP and Syslog.
- Universal redundant power supply 100-240 V. 50/60 Hz. 50 VA.
- No mechanical fan. Silent. Natural convection cooling.
- Operating temperature 0 to 45º C.
- Weight approx. 5 kg (11 lbs).
- Width 482 mm (19”), height 1 rack unit, 44.5 mm (1.75”), depth 300 mm (12”).

SYSTELSET+ IP phone programmed with control application

- 7” multi-touch screen.
- 8 fully programmable function keys.
- Programmed with the SYSTELSET+ application on Android 5.1.1.
- 12-key telephone keypad.
- Dual Gigabit Ethernet port: 10/100/1000 Mbps.
- Headset connector: 1 x RJ9 (4P4C).
- Handset connector: 1 x RJ9 (4P4C).
- USB 2.0 port for USB headsets, wired or wireless.
- HD Voice.
- Speakerphone.
- External power supply 100-240 VAC 5V DC and PoE (IEEE 802.3af), class 3 max 6W.
- Dimensions (W* D* H* T): 259.4mm * 235.2mm * 194.5mm * 42.6mm. Weight 916 g.

August 2025. Specifications subject to future updates. Download the latest version at www.aeq.es,

CAT.SYSTELMAX.2509

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