HD VoIP ON AIR Telephone System

Dante SPOKEN HERE /AES67

Systel IP

Talk-show and multiconference system for broadcast and business environments

Operating Environment

Field of application of the product

SYSTEL IP 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 IP 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:

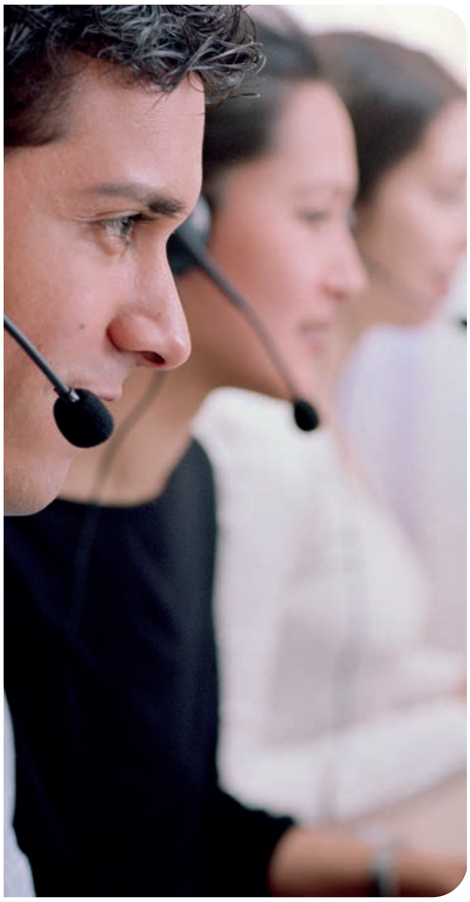
In 1994 AEQ developed the Systel 3000 conference system with control for digital telephone hybrids on conventional telephone lines in console multiplex format.

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.

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 IP 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 IP does not operate on hybrids, but on a 4-wire digital matrix: all the lines can intervene live and simultaneously without loss of quality.

- Significant cost savings can be obtained by connecting the entire system to an Internet telephony provider, or as extensions of the IP PBX that is already in service in the corporation.

- SYSTEL IP16 shares the IP lines in a very flexible and dynamic way with up to 4 studios through very simple analog or digital cabling not having to deploy special and expensive audio nodes. SYSTEL IP 16, also offers many channels of local audio input and output through IP, in Dante format, compatible with AES 67.

- It is possible to create much larger installation, with dozens of studios or even for a whole network of broadcasting stations. In such scenario, each SYSTEL IP16 will be a simple set of extensions and that can share phonebooks and users.

- SYSTEL IP control terminals are extremely powerful, flexible, economical and practical. We can use:

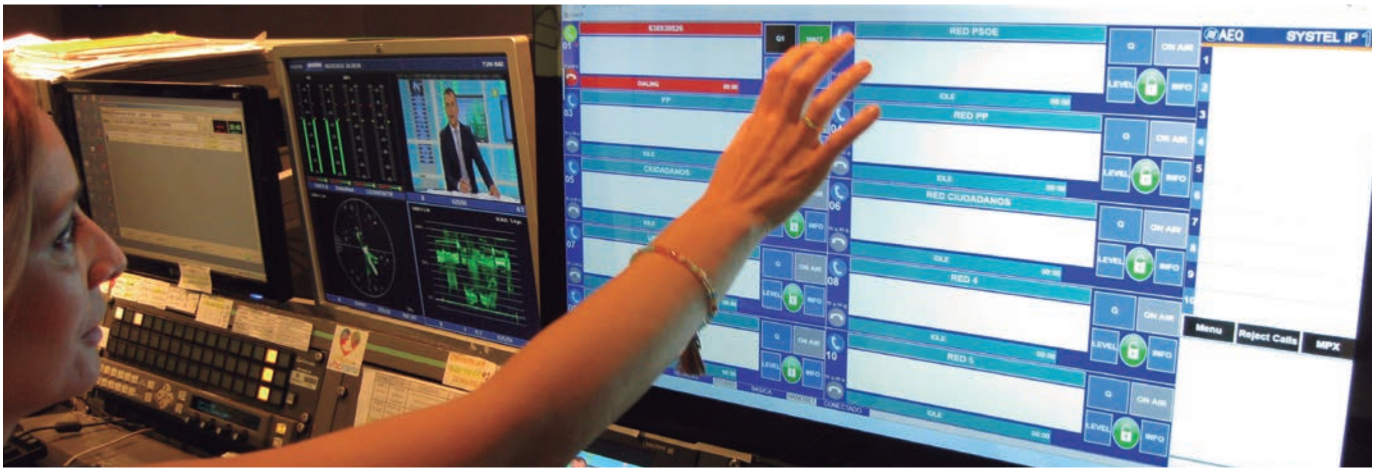
  - An IP phone in order to make calls and talk to the remote party, as well as a software application that can be installed on any PC.

  - An IP phone with touch screen, in order to make calls and talk to the remote party, running a specific embedded 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

SYSTEL IP 16. The heart of the system is a 19" rack format equipment:

For 16 IP phone lines, 4 additional lines for IP operator phones, 4 digital inputs/outputs, 2 analog inputs/outputs and 32 Dante protocol IP inputs/outputs, enough for up to 4 studios.

The unit behaves like multi-line IP Phone with SIP signalling protocol. Compatible with IP PBX, SIP Trunking and virtual PBX. Analogue and ISDN lines supported through gateways.

Coding algorithms include the proper ones in telephony: G726, G729 and G711. Also incorporates G722 coding with extended bandwidth to 7 kHz, which characterize it as "HD" and makes it compatible with N / ACIP AudioCodecs and SIP-Phones (Any AEQ Phoenix AudioCodec and most PC telephony software).

Control Applications

CONFIGURATION AND OPERATION SOFTWARE

SYSTEL IP 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.

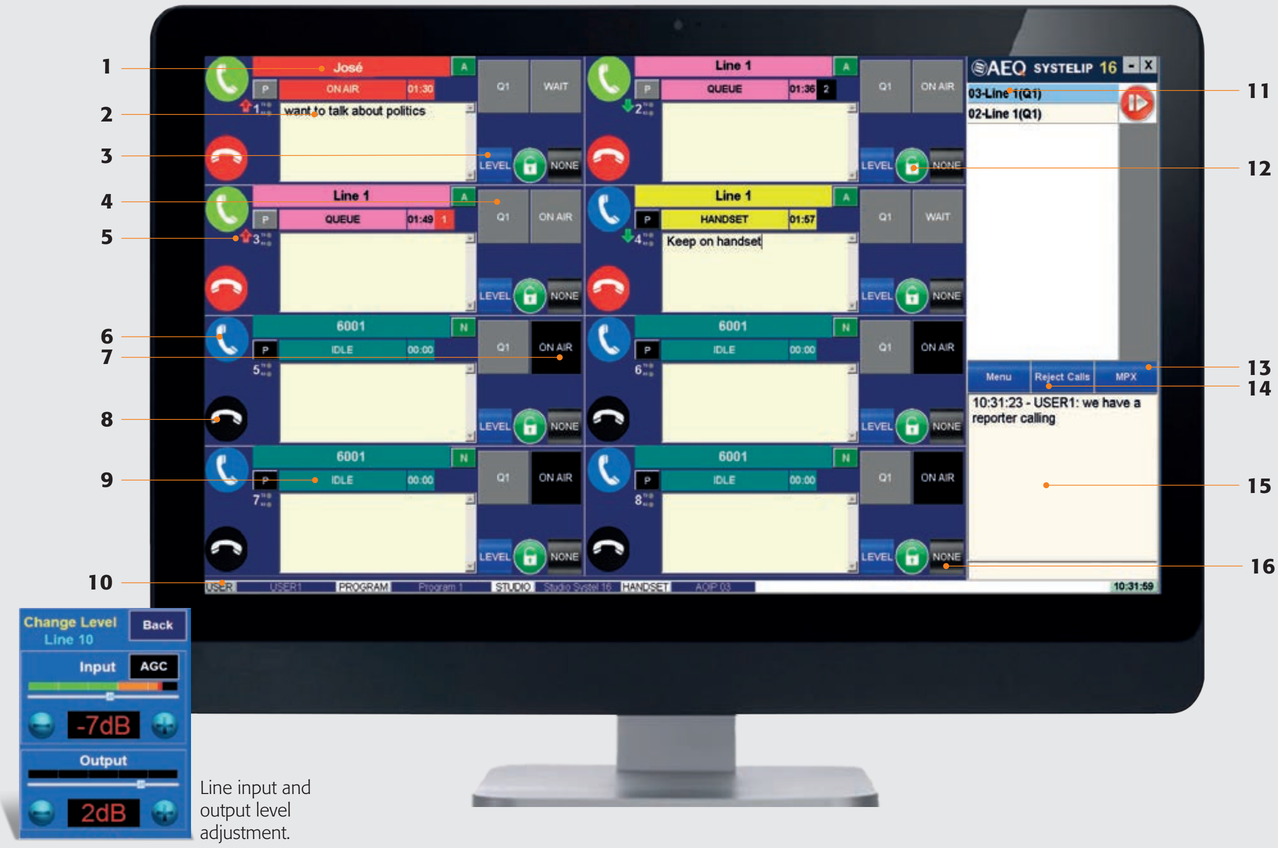
SYSTELSET+:
Integrated within the SYSTELSET+ divece, allows for a very flexible operation while avoiding the need for a PC in the control rooms or other reduced spaces.

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.

NEOGROUPE SUITE :
The NeoScreener, NeoWinner and Neoagent applications from Neogroupe, integrated with Systel, allow for great agility in complex TalkShow and are very popular in USA, France and other countries.

SYSTEL IP ORIGINAL

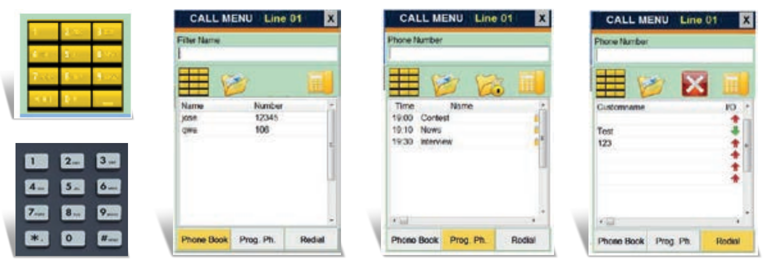
This is the best option in high-productivity environments where functionality is distributed in roles such as producers, controllers and talents. Calls are dialled or accepted, put on hold or pre-listened, their send and return levels can be adjusted, they can be diverted to auxiliary circuits, put on 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.



Line input and output level adjustment.

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.



1 2 3 4

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 SYSTELSET+'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 (without removing the previous ones).

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

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.

SYSTEL IP TV

Eases flexible and generic control of Systel IP 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:

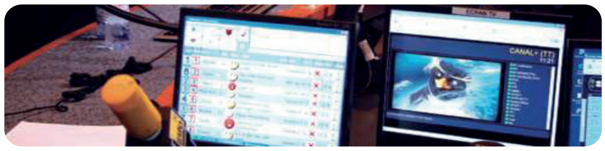
- Accept incoming calls manually or automatically, label calls, put them on air or leave them in a multi-conference group.
- Leave calls listening to their assigned (N-1) feedback.
- Put the calls on air, routing them to the assigned internal audio circuit.
- Leave calls in different multi-conference 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.

Systel IP TV Screen

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.

NEOGROUPE CONTROL SYSTEM

Systel IP 16 can also be controlled using Neogroupe developer's Neoscreener and Neowinner applications, taking advantage from the exclusive features of these products:



NeoScreener. Is a software application designed to manage complex Talk Shows with special screens for each different radio and television work flows, including link to NeoWinners / NeoAgent:

- User friendly interface.
- Numerous features.
- Designed for touch screens.
- Full lines control.
- Caller identification (CLID).
- Chat text conversation between workstations (screener/talent).
- Unwanted callers management.
- Calls priorities.
- Strong database with search fields and reports.

NeoWinners / NeoAgent. Applications are designed to manage quizzes and award ceremonies using IOS applications, webpages, SMS, e-mails as well as Talk Shows.

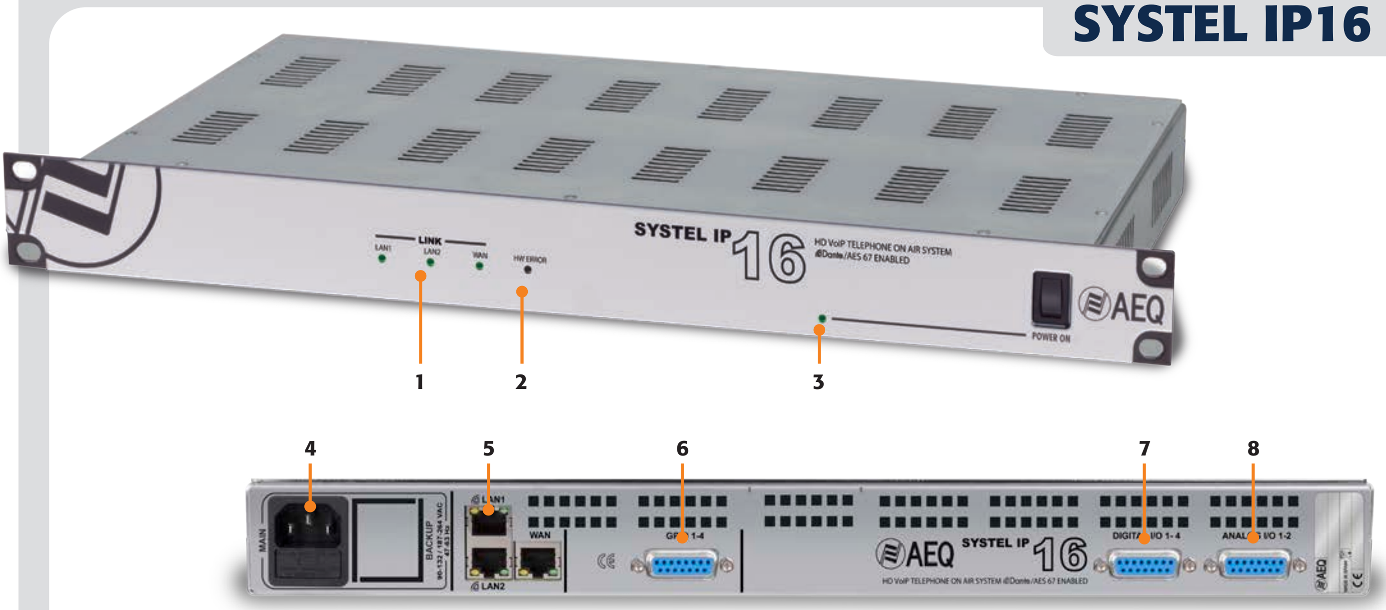
NeoScreener Screen

Talent Screen

NeoWinners Screen

PRODUCT DETAILS

SYSTEL IP16



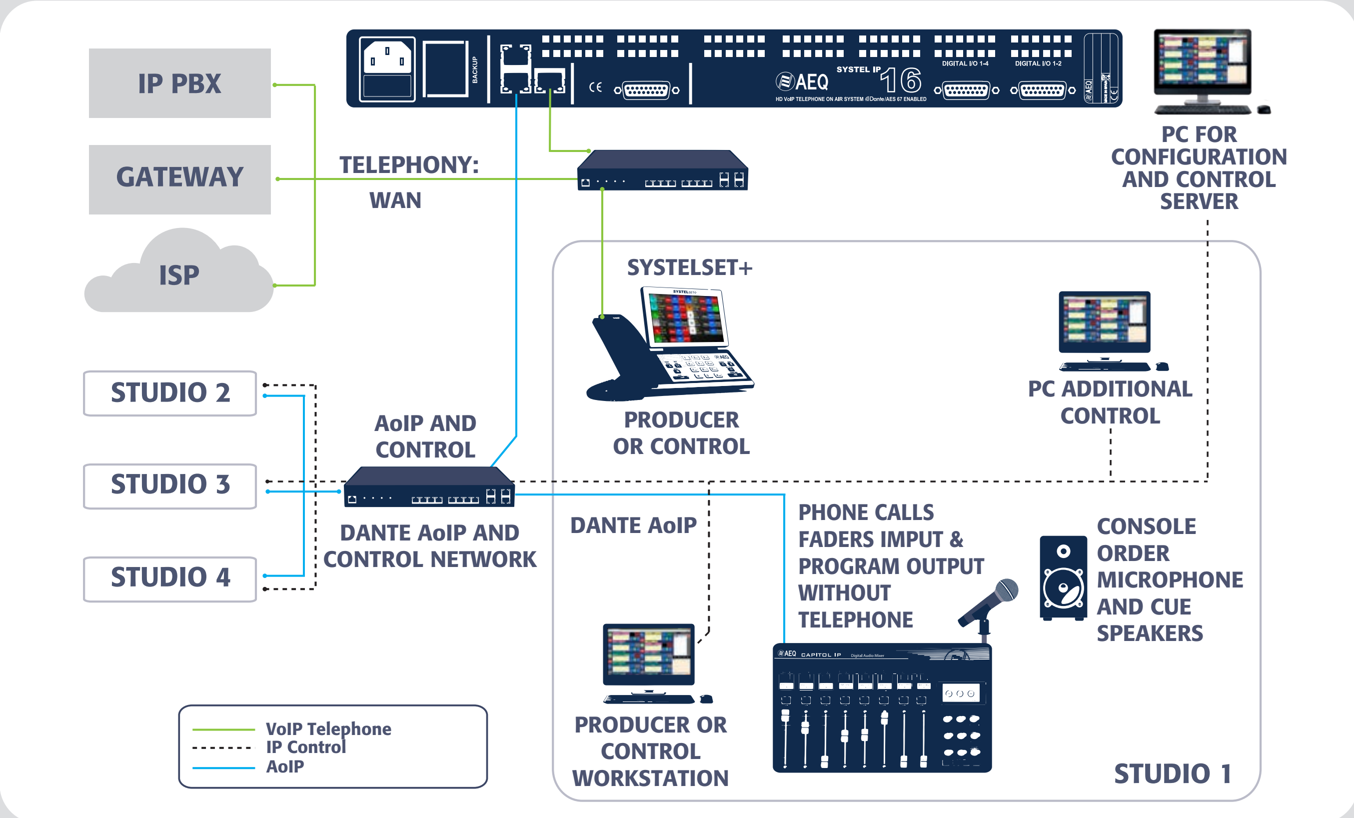
1 Link Indicators.	**3** Power Indicator.	**5** IP Network Connectors.	**7** Digital Audio Connectors.
2 HW Error Indicador.	**4** Power Connectors.	**6** GPIO Connectors.	**8** Analog Audio Connectors.

SYSTEL IP 16 and SYSTELSET+ control phone terminals “for four” studios

The SYSTEL IP 16 behaves as a combination of 16 IP telephone terminals that can be shared in a flexible and dynamic way between the four studios. Calls arrive by the WAN Ethernet connection The control is provided through the Ethernet LAN by a PC that has the control and configuration applications installed. Audio can be connected locally using analogue or digital connections, or through the Dante network. Control terminals access the control server and Systel equipment via IP.

SYSTEL IP16 allows the use of 4 IP phone terminals for control (one per studio, as an example). There are two analogue audio inputs and outputs, and 2 dual digital ones in the device, what allows for audio connection without requiring Dante. The control room technician can attend calls by talking through order circuit and listening y CUE if the SYSTELSET+ control terminals are assigned to the producers. An auxiliary bus without telephone audio is sent from each studio console in order to be added to the rest of phone calls’ audio and provide particularized return to each telephone.

Additional control PCs can be installed.



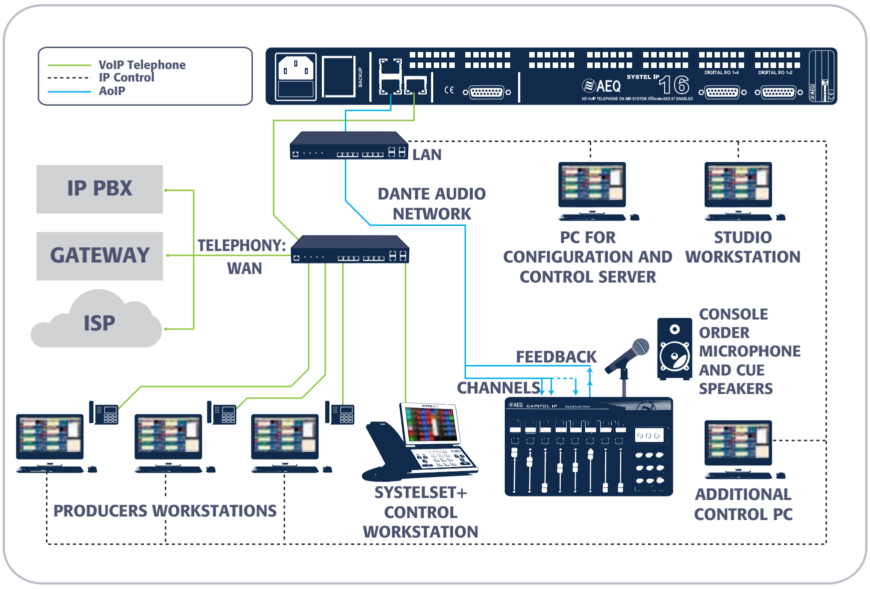
SYSTEL IP16 in a studio with producers and several audios to console

The device’s IP connectivity, with up to 32 incoming and outgoing channels, allows for differentiated audio to be sent in 1, 2, 4, 6 or more console faders.

From the mixing console an aux bus is sent in order to add the rest of the telephone lines and to be able to provide customized return for each caller.

We count with 16 IP external phone lines plus 4 internal ones for operators and producers. Producers use conventional IP phones and PC to label calls and chat with the booth.

The control technician uses a SYSTELSET+ terminal and, when chat following is required, the control application can also be installed on one of the control PCs.



SYSTEL IP 16, a user report for the Systel IP TV application together with an intercom system for the external coordination of a TV production center.

TV transmission centers generally have several studios. This particular case is involving 4 studios. A sound mixer and several intercom user panels are located in each studio.

The Systel IP TV software is running on a PC and as an integrated solution for control of each studio as well as an IP phone for communication purposes. The main usage of the equipment is to making calls and coordinating the communications of each studio.

There is also, an intercom matrix, a Systel IP 16 device and optionally a PBX with handsets shared by the 4 studios.

There are two IP networks installed: the internal one or LAN, and the external one (WAN). The Systel IP primary LAN port has to be connected to the LAN switch. The intercom matrix, audio mixing consoles, the intercom user panels and the control PC are connected to the same switch. The Systel IP 16 WAN port is then connected to the WAN switch, together with the IP phones and the ISP phone access. The station's PBX can also be connected to the WAN switch; this may be especially useful when we want to receive calls from the station's regular PBX handsets or if there are more than one Systel IP 16 is installed.

16 bi-directional audio channels are established between each Systel IP 16 and the Intercom matrix. This matrix will create routings with the user panels and the audio mixing consoles, as defined in the Intercom control applications Crossmapper and Live Crossmapper, establishing work groups, Party Lines or other more complex configurations if required.

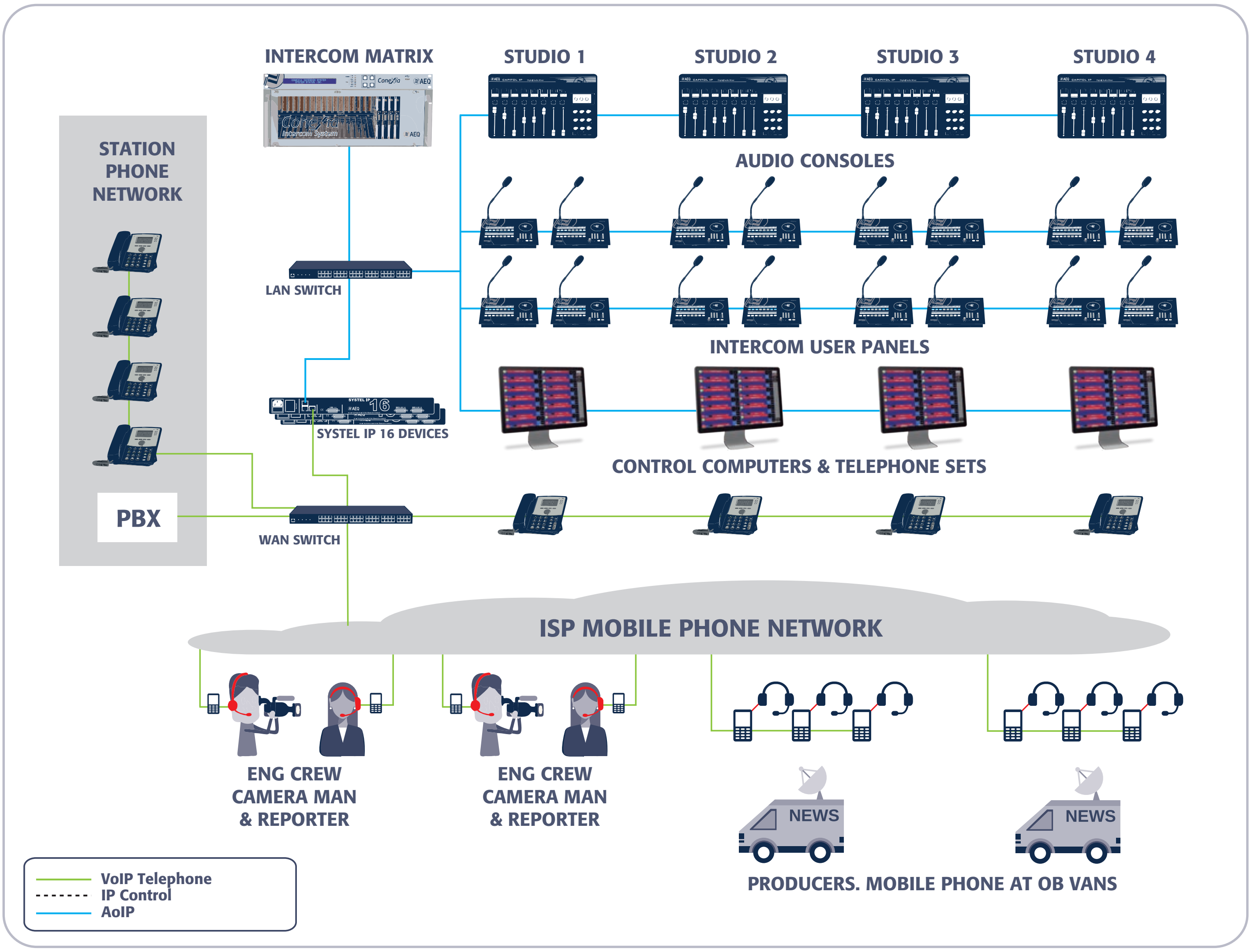
The 16 Systel IP lines can be dynamically assigned to the studios and calls can be sent and received using the Systel IP TV software. The associated telephone set can be used to dial and also to speak before the communication is established. Calls can be received from outside and, if automatic answering has been configured, the call enters the matrix and, from there, it goes to the assigned user panels. Also, pre-programmed calls can be defined, assigned to particular keys in the different user panels.

The intercom matrix establishes bidirectional audio routes with the console and complex, party-line based, routings with the different user panels as a function to each particular requirement for production.

This particular application provides coordination with ENG crews through mobile phones. The journalist will receive program return and commands in his/her earpiece. The cameramen communicate using 4-wire circuits with a party-line group previously set-up for the intercom Matrix, through headset combinations connected to their cell phones.

When operating with a mobile unit, at least three 4-wire circuits are established and integrated into a specific intercom group.

Also, the Systel IP 16 allows for coordination between the different mobile devices, allowing the studio to dynamically determine who speaks to whom and at any given instant. Further, it can be decided who is receiving auxiliary signals, such as program return.



EXCLUSIVE COMPONENTS

"Engine" para 16 líneas IP: SYSTEL IP 16

All the processing and connectivity power required to manage 16 IP lines is concentrated on a frame with 1U rack height: IP connectors for audio, control and voice, 2 analog inputs and outputs, 4 AES3 digital inputs and outputs, 32 Dante / AES 67 inputs and outputs, 12 GPI and 12 GPO. Includes configuration and user software for an unlimited number of terminals.

Wiring Accessory: FR CAB INP

DB15 male connector to four unscreened balanced pairs of a 4 meter cable (other end "open-end"), to facilitate the wiring of 2 inputs and 2 audio outputs in SYSTEL IP 16.

Wiring Accessory CAB FR GPIO

DB15 male connector to 4 meter cable (other end "open-end"), to facilitate the wiring of 2 GPI and 2 GPO in SYSTEL IP 4 (max. one needed per unit) or SYSTEL IP 12 and SYSTEL IP 16 (max. three needed per unit).

SYSTELSET+ Handset

Control terminal based on an IP phone with touch screen running a new configurable control application which can be adapted to the most varied operating ways. The need for a PC on each work place for full functionality is avoided.

ANCILLARY COMPONENTS

In order to setup a System IP system in a particular environment, you may need to add some additional computing or telephone devices that are readily available on the market or even already installed in any office or radio / TV station. The requirements are quite relaxed, but in case you would require, AEQ can recommend or even provide those complements, certified by our System Engineering department.

Switch Ethernet

The equipment is connected to a network for control (LAN) and another one for VoIP (WAN). If not already created for other purposes, an Ethernet switch needs to be installed for each network. Both networks can be unified in smaller installations.

Configuration, control & databases PC

Almost any PC able to run Window XP or above is adequate for the installation of the configuration and control applications. It is however very convenient that the PC where the control application is installed has a tactile screen. All in One type devices are well suited. Any PC can be used for the data bases. For larger installations, dedicated or shared server can be used. All the applications can run on the same PC in small installations.

Operator's micro-headphone combination

Some producers that are continuously receiving calls need the capacity to connect operator headsets. Both wired and wireless units can be found in the market. They must have a RJ-9 connector, in order to be connected to a substitute to the SYSTELSET+ handset which must be previously disconnected. Some wireless headsets also provide an auxiliary connector in order to be able to have both handset and headset connected.

POTS or PSTN FXO Gateway

Converts conventional telephone lines into IP lines. There are different models available.

ISDN Gateway

Converts conventional ISDN lines into IP lines. There are models for 1, 2 and 4 basic (BRI) or primary (PRI) access.

IP PBX

SYSTEL IP doesn't have to rely on a PBX, but it should receive the IP calls from somewhere: Gateway, SIP Trunking or IP PBX. Therefore, if you take the opportunity to totally migrate your station's telephony to IP we can help you to select the IP PBX that suits your needs.

IP Handset

SYSTEL IP allows compatible IP phones to be used as handsets instead of SYSTELSET+. As an advantage, they allow direct dialling through their physical keyboard instead of using the application. SYSTEL IP 16 holds sufficient internal IP addresses for IP handset connections. The use of the SYSTELSET+ as an IP Handset is really advantageous since it also provides the SYSTEL IP Control Application.

IP Phones

SYSTEL IP 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

GENERAL FEATURES

SIP communications protocol: compatible with VoIP trunkings, free PBX, SIP Phones such as Phoenix Pocket or Phoenix Lite, N/ACIP compliant Audiocodecs such as Phoenix Mercury, Phoenix Studio, Phoenix Venus or Phoenix ALIO and POTS, ISDN, E1 and T1 FXO.

Based on non-blocking digital switching matrix, all16 lines can be simultaneously live participating in a program with no loss of quality.

GPI/Os: 4 GPI, 4 GPO and power supply on each DB15 female connector. All functions are replicated over TCP / IP in the control network.

Audio specifications
- Analog inputs: input impedance: 20Kohm. Electronically balanced, professional line level.
- Nominal input level: +4 dBu. Max. input level: +24 dBu.
- Analog outputs: output impedance < 100 ohm. Electronically balanced, professional line level.
- Nominal output level: +4 dBu. Max. output level: +24 dBu.
- Digital inputs / outputs: AES / EBU interfaces, configurable as AES-3 or SPDIF. Inputs include SRC.
- AES 1 input can be used for external AES-11 synchronization.

There are also Dante, AES 67-compatible inputs and outputs. Dual IP LAN interface compatible with Dante native redundancy. Synchronization is transported through the network.

Audio processing
- Phone audio in G.711, G.726, G.729, 50Hz - 3KHz.
- High-Definition audio with G.722 algorithm: 50Hz – 7KHz.
- Echo cancellation. Automatic gain control.
- Independent, digital gain control for all inputs and outputs with an adjustment range of +/- 12 dB and muting.
- Automatic gain control for telephone returns.

Configuration software and control server

Windows 32 and 64 bit operating systems: Windows XP, Windows Vista, Windows 7, 8 and 10.

Functionality (can be individually configured for each user group).
- Assigns audio, IP phone and chat circuits to the different studios, univocally.
- Renames circuits.
- Defines and manages phone books, allowing the user to share, edit and copy them.
- Defines PFL signals assigned to each studio.
- Defines auxiliary and master signals assigned to each studio.
- Configure the initial audio levels for each line and each study.
- Configures the format of the client screens, defining the number of lines per program, console operation, and the use of one or two call queues.
- SIP configuration for communication with an IP PBX, FXO gateway and external (Internet) or internal (LAN or WAN) service providers.
- Distinguish and protect with rights on activities the functionalities of different types of user.

SYSTEL IP Original and SYSTEL IP TV control clients

Windows 32 and 64 bit Operating Systems: Windows XP, Windows Vista, Windows 7, 8 and 10.

Functionality (can be individually configured for each user group)
- Call establishment: by number dialling, with SIP identifiers, or from phone book entries.
- Call establishment: by dialing IP phone numbers taking advantage of the IP handset functionality.
- Optical tally and acoustic RING signal.
- Caller ID. Contact list matching. Adding of a temporary name.
- Pick up incoming calls manually and automatically.
- Define and manage phone books, either general or private to each program.
- Create and manage phone call schedules.
- Queue the on-air ready calls on one or several faders, allowing for their re-ordering and dynamically checking them.
- Grant the VIP attribute to a call in order to keep it on a dedicated fader.
- Accept incoming calls, either manual or automatically.
- Register new contacts in the call book.
- Talk by means of the headset or microphone / headphone with the people at the remote line end.
- Put calls on hold, while the caller can listen to the program.
- Put calls on-air so they can contribute to the program.
- Place several call on CUE on one or several Faders calls ready to be placed on-air, allowing for the dynamic reordering of calls and TalkBack.
- Tag a call as VIP, thus making it exclusive to a fader.
- Changes input and return levels for every phone line in the studio.
- Changes input and return levels for every phone line in the studio and every headphone.
- Display the status of all the phone lines and where they are being routed to.
- Label calls. Chat among the different controllers assigned to the program (Only Original version). Black-list management. Call-barring.

SYSTELSET+ Control client

Android Operating system.

Functionality (configurable for each users group)
- Making calls: dialing numbers, SIP identifiers or register in phone books and schedules.
- Making calls: dialing telephone numbers.
- Sending an optical and acoustic RING signal.
- Showing the caller's ID or number, substituted by the local name in the phone book or a temporary name assigned from SYSTEL IP Classic application.
- Answering incoming calls automatically or manually.
- Registering new contacts in the phone book.
- Managing general and program-private phone phone books.
- Managing phone book schedules.
- Talking by means of micro-earphone, hands-free or micro-headphone with the person at the other end of the line.
- Leaving calls on-hold while listening to the program.
- Putting calls on air so they can contribute to the program.
- Queuing on one or several faders the calls that are ready to be put on air, allowing for its re-ordering and dynamic consulting them.
- Assigning the VIP attribute to a call so it can be kept on a dedicated fader.
- Changing the phone listening levels and input and return levels for each of the studio's phone lines.
- Displaying the status of each phone lines and where are they being routed, while they are kept in that status.
- Distinguishing and protecting with rights the functionalities assigned to producers, operators and presenters.
- Managing black lists and incoming call barring.
- Working with typical European and USA naming and functions.
- Activating the Dump Mode, in order to hang up or not calls after they have been on-air.
- Activating Page Lines, in order to send warnings to all lines and receive simultaneous replies.
- Activating Auto Next, that puts the next call on-air after one has been hung up.
- Activating PickUp Incoming in order to automatically connect to the oldest call when picking up the handset.
- Activating AutoQueue, in order to automatically queue the call when hanging the handset.
- Activating Direct Dial, in order to skip the "select line" step to make a call.
- Activating Direct Next, that puts on air the calls even if they have not been attended previously.

SYSTEL IP16 "16-lines IP engine"

Inputs and outputs
- DB15 female audio multi-connectors. Two I/O each.
- 2 analog balanced inputs.
- 2 analog balanced outputs.
- 2 digital AES- EBU (AES3 or SPDIF) dual inputs.
- 2 digital AES- EBU (AES3 or SPDIF) dual output.
- 1 WAN IP port for 16 VoIP lines, plus 4 VoIP lines for control phones.
- 2 LAN IP ports for control and 32 AoIP inputs / outputs in redundant Dante / AES-67 format.
- 1 DB15 connector for 4 opto-coupled GPI and 4 GPO each one.

General characteristics
- Power supply. Universal 100-240 V. 50/60 Hz. 50 VA power supply.
- Silent operation: natural convection cooling.
- Weight: 4 Kg (8,8 lbs).
- Width: 482 mm (19") 1U rack height = 44 mm. (1,75").
- Depth: 356 mm. (14").

SYSTELSET+ IP phone with preloaded control application

- 7" multi-touch screen.
- 8 pre-programmed function keys.
- SYSTELSET+ pre-loaded, running on Android 5.1.1.
- 12-key phone keyboard.
- Dual Gigabit Ethernet port: 10/100/1000Mbps.
- Headphone connector: 1 x RJ9 (4P4C).
- Handset connector: 1 x RJ9 (4P4C).
- USB 2.0 port for wireless or USB headphones.
- HD Voice.
- Hands-free.
- 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 gr.

March 2019. Specifications subject to changes without prior notice. Download the latest version www.aeq.es, www.aeq.eu, or www.aeqbroadcast.com.

CAT.SYSTEL.21_01

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