

Studer Call Management System

Product Information



The Studer Call Management System uses Voice Over Internet Protocol technology to replace the physical Telephone Hybrids and Codecs. The control and signalling systems of the hybrids and codecs are integrated parts of CMS. It will interface to most PBX's with VOIP interfaces, direct with ISDN lines or to the Internet. It also allows the control of analogue lines (POTS). When interfaced

with the Studer OnAir 3000 console platform, the result is better, more effective and intuitive handling of outside sources. The CMS has applications for different situations where telephone and/or codecs are involved in a broadcaster's daily work, ranging from simple off-air interviews to large game shows.

Studer Call Management System (CMS)

At the heart of the system is the CMS server that can handle the entire gamut of telecommunications formats, from analogue POTS circuits, through ISDN, to the latest voice-over-IP technology.

The system's main job is to automatically manage and distribute calls to clients used by reporters and operators working in studios and editorial offices.

The entire system is under straightforward software control, managed by a client application that is installable on any number of networked PCs. Users do not have to be concerned with distinctions between analogue, ISDN, or VoIP telephone traffic, or indeed a mix of all three.

Voice-over-IP: The Studer CMS handles VoIP telephone traffic with ease. With Ethernet connections, the voice server can control and distribute up to 240 channels either directly or via holding areas to broadcast studios, editors' desks, or call operators. There are virtually no limits to call patching and management via holding areas. Standard audio cards provide a two-way interface with the audio world.

ISDN: Up to 120 channels can be connected with the voice server via an ISDN adapter and managed/distributed using voice-over-IP. Again, operation is via the same client that controls VoIP and analogue telephone lines.

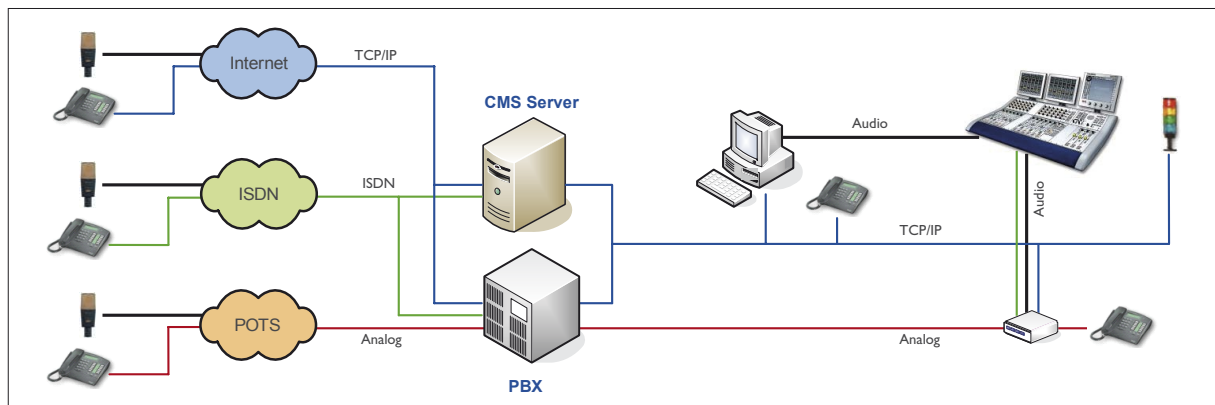
POTS: The Studer CMS can of course handle calls on analogue telephone lines. In this scenario, the server manages up to four analogue telephone hybrids per studio or audio workstation via network programmable control units that are operated via the same software client.

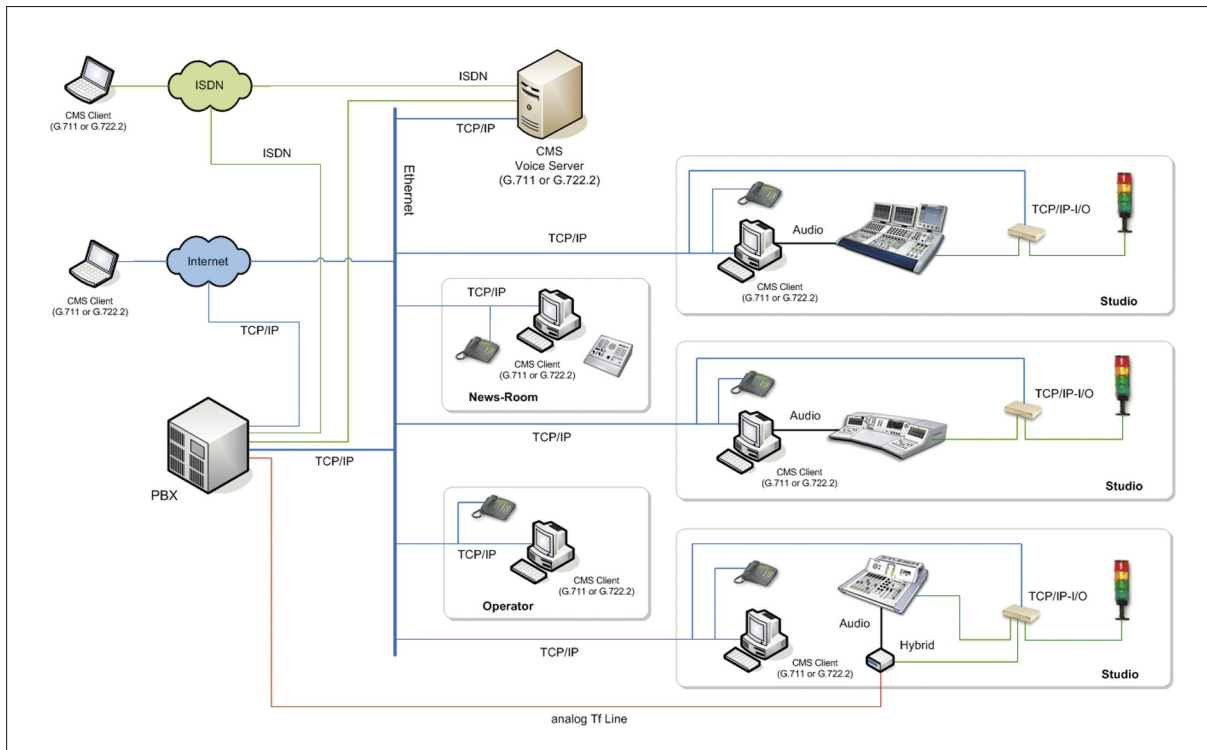
Not only does the Studer Call Management System provide virtually unlimited control over telephone traffic, it also handles related peripherals like signal lamps, studio red-light installations, and other items of broadcast equipment that need to be controlled in tandem with communication systems. Connected to a telephone network that supports caller ID (e.g. ISDN), the Studer Call Management System can also be linked with directory services. This allows focused analysis of caller demographics, which holds major benefits for gaming and voting applications.

System Overview

The Studer Call Management System is basically scalable from very small systems (CMS Server + few clients) up to very large systems (Multiple CMS servers with hundred of clients). The system can either be connected directly to ISDN and Internet (VoIP) or communicate with a PBX (Private Branch Exchange) or a VoIP exchange system (like Cisco Call Manager or others). Redundant operation of CMS Servers is possible.

Communication (control, voice/audio and signalling) between a CMS Server and CMS client is only via TCP/IP. The CMS offers the following standard communication protocols:
 H.323 for VoIP
 G.711 softcodec for speech coding
 G.722, G722.2 available soon, Speex, Ogg Vorbis for high-quality audio.
 (-> 16 bit, > 20 kHz)





OnAir 3000 Integration

The CMS client can be integrated to the Faderscreen (touch) of the Studer OnAir 3000 mixing console or can run on its own screen.

The CMS Client and the OnAir 3000 mixing console can communicate together. It is possible to serve the caller on the CMS Client as well as on the mixing console (OnAir / Hold switch, Talkback, PFL).

Server/Client structure

The Studer Call Management System is based on the CMS Server and CMS Clients that are available for various applications. The heart of the system is the CMS Server software and the XCAPI interface.

XCAPI provides telco services for VoIP (Voice over IP) as CAPI does for ISDN. Various software modules of the CMS Server software manage and distribute incoming and outgoing communication lines connecting them to the clients and/or to other VoIP systems (like Cisco Call Manager).

The number of CMS clients per server is unlimited, the maximum number of simultaneous lines is 240 for VoIP and 120 for ISDN lines.

Using a CMS Reporter client with wideband softcodec allows high-quality audio transmission from outside right to the mixing desk in the studio using the same user-interface as for phone calls.

The CMS allows extensive signalling of calls over TCP/IP with relay interfaces.

The Studer Call Management System uses only standard IT equipment and components and supports almost any audio-card for coding/decoding of audio from/to the audio equipment like a mixing desk. Each CMS client can support up to 4 audio-channels (each with clean-feed), either with 2-channel or multi-channel boards.

Applications

The CMS can cover a wide range of communication related applications in a broadcast house. Its flexibility and scalability allows systems with only a few clients up to complex installations with hundreds of clients in studios, newsrooms and outside broadcasting surroundings.

Telephone system:

The CMS is a full blown telephone system to interface communication lines (POTS, ISDN, VoIP) with professional broadcast equipment like an on-air or production mixing console with a single GUI. It can replace any telephone- or ISDN-hybrid solution and additionally offers VoIP.

Audio routing:

The CMS is able to route high-quality audio (G.722) over standard digital communications lines (ISDN, VoIP) with the same GUI mentioned above. The use of softcodecs avoids expensive hardware solutions and allows compact mobile equipment.

Intercom:

The CMS is a complete communication system (Softphone) for internal and external connections.

Voice Mailbox:

The CMS offers a highly sophisticated answering system on multiple levels with automatic recording.

Gaming/Voting:

The CMS allows extensive on-air gaming and voting with online analysis. The system allows even to filter callers with automatic Q&A cascades on different levels until they finally reach the holding areas in the studio.

Marketing-Tool:

With the CMS it is possible to analyse caller demographics (with graphical output) to a very high level.

CMS Clients

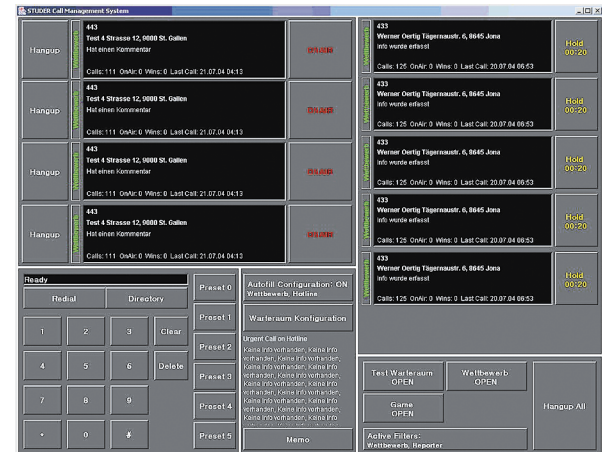
The Studer Call Management System (CMS) offers four types of clients for best operation in various applications:

- Studio Client
- Newsroom Client
- Reporter Client
- Operator Client

Studio Client

The Studio Client is designed for operation in on-air and/or production surroundings and offers a wide range of functionality to manage calls either directly or forwarded from other CMS Clients.

- optimized for touch screen
- priorities for holding areas
- filter functionality
- AutoFill function
- Softphone integrated
- info window
- VIP signalisation
- Integrated with OnAir 3000

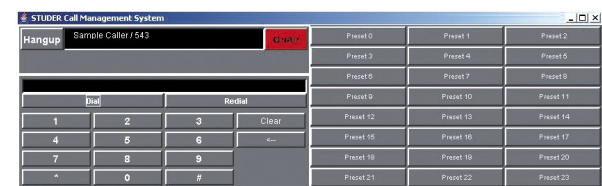


Newsroom Client

The Newsroom Client is for the journalists on the news desk who are mainly conducting off-air telephone interviews.

Newsroom Client features:

- Integrated soft phone
- Two outgoing calls simultaneously
- Many preset buttons
- Using only a small window so other applications can be accessed simultaneously

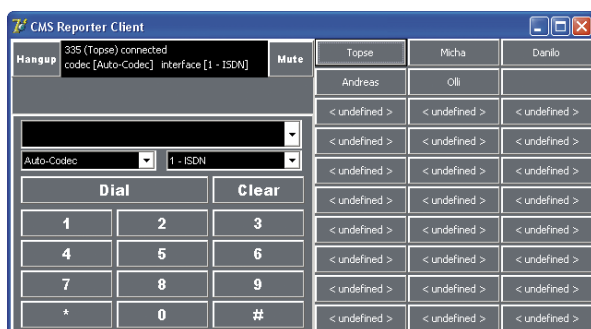


Reporter Client

The Reporter Client is a soft phone that can be used independent of the CMS. The client is meant for the reporter working externally and will be compliant with most codec standards, i.e. G.711, G.722 ...

Reporter Client features:

- Many codec standards built in
- Independent of CMS
- Can communicate VOIP over ISDN



Operator Client

The Operator Client is primarily for the telephone operator answering/screening the calls. The operator will then add the caller's data to the information window and pass the call to the studio or control room.

Feature	Studio	Newsroom	Reporter	Operator
Multi-Language (Uni-Code)	✓	✓	✓	✓
OnAir 3000 integration	✓			
Caller Identification				
Name/address of caller	✓	✓		✓
Editable additional info of the caller (for all calls and/or for current call)	✓	✓		✓
Time the caller is waiting	✓			✓
Number of calls of the caller, date of last call	✓			✓
Number of on-air's of the caller, date of last on-air	✓			✓
Number of wins of the caller, date of last win	✓			✓
Integrated Softphone				
DTMF send	✓	✓	✓	✓
Redial	✓	✓	✓	✓
Presets	✓	✓	✓	✓
Shortcut keys (considering Softcodec)	✓	✓	✓	✓
Holding areas				
Color for each holding area	✓			✓
Prioritizing of holding area	✓			✓
Direct on-air of callers in holding areas	✓			
Moving of callers from one holding area to another	✓	✓		✓
Forwarding of callers to other CMS clients	✓	✓		✓
Hang-up for each caller or all callers	✓			✓
Winner-Notification	✓	✓		✓

XCAPI

For a long time now, ISDN has been the relevant standard for modern telecommunication. Strong services and high quality of speech are setting the benchmarks for any competing technique.

Many businesses nowadays are using ISDN based software solutions compatible to CAPI 2.0 for better customer service and a more efficient performance. The scale includes simple fax-servers, CTI- and IVR- products up to complex UM-solutions, —or routing solutions like an ACD. Most of these products are connected to a standard PBX via an ISDN card based on CAPI 2.0.

But parallel to this, a new standard has already begun to establish itself as a new professional reference: Voice over IP. VoIP uses the normal Ethernet computer network, originally designed to be used for data communications. VoIP makes use of this network to transfer normal conversation, too.

Until now integrating VoIP meant difficulties in many respects: a complex integration scenario, less quality of service, low quality in speech transmission. The simple reason for this is that ISDN cards are not compatible with VoIP. Therefore expensive gateways had to be installed, to allow existing ISDN applications and hardware to integrate telephones and other equipment designed for the VoIP net.

XCAPI — a new product from German telecom software specialist TE-Systems — offers an entirely different solution with most positive results: Installation is totally easy, integration is surprisingly wide-scaled. Hence XACPI can be used as a plug'n'play replacement for a standard ISDN-card, allowing all applications and hardware to access both the ISDN and the VoIP net.

Any application identifies XCAPI as a driver for an ISDN-card, offering fax and supplementary services and also conferencing. Instead of using an S0 or S2M port, XCAPI connects to the telephone system via the Ethernet port to communicate according to H.323.

XCAPI's perfect quality of speech is a result of the brand new X-Enhance technology, developed by TE-SYSTEMS. It equals the clarity and speed of ISDN and VoIP for best communicational results.

Fulfilling all relevant standards is a key issue for XCAPI. All applications and hardware compatible to the international norm CAPI 2.0 can easily be connected. For communicating through the data network XCAPI is referring to the ITU standard H.323. Any fax or speech data from corresponding sources XCAPI transfers with best reliability from one point to the other.

On each machine XCAPI operates up to 240 lines. The capacity can be scaled freely according to the required bandwidth. Anything is possible — from two up to some hundred lines by clustering machines.

High flexibility and adaptability offer XCAPI a wide range of possible operations. With XACPI existing CAPI 2.0 applications get their step into the future. XCAPI interprets incoming DTMF tones and can produce them itself. XCAPI distributes calls, sends and receives fax and operates conferences, even if their participants are using both ISDN and VoIP technique.

Apart from this XCAPI offers the integration of existing ISDN-cards in its operations. Hence mixed infrastructures can be integrated in the most easy way - simply by installing the new middleware for VoIP and ISDN!

XCAPI operates as a driver in the kernel mode and is a middleware for the integration of VoIP and ISDN. The graphic shows its operation scheme and widespread connectivity.

XCAPI - Facts And Data

XCAPI is the new middleware for VoIP and ISDN and it supports many standards, thus opening new doors to applications, giving them better performance and easier integration. The following list shows some of the major technical features:

CAPI 2.0

XCAPI conforms perfectly with CAPI 2.0. It supports all necessary B-channel protocols for speech:

- B1: 64kbit Bittransparent, T.30
- B2: Transparent, T.30
- B3: Transparent, T.30

Fax support (T.30) is offered for:

- MH
- MR
- MMR
- ECM (Error Correction)

Furthermore XCAPI supports functions needed for the use with unified messaging applications. Presently these are:

- Recognition and generation of DTMF-tones (outband)
- Recognition and generation of DTMF-tones (inband)
- Line-Interconnect.

VoIP Abilities

XCAPI offers outstanding functionality for VoIP. Among others it is highly compatible with various products based on H.323. XCAPI gives interoperability with products and standards of:

- Cisco
- Innovaphone
- Radvision
- MS Netmeeting
- Snom
- Symbol
- OpenH323
- and many more

Presently XCAPI supports the following codecs for speech:

- G.711 alaw / μ Law
- GSM 6.10
- Speex
- G.722.2 (soon available)

Amongst others the following standards for VoIP are supported by XCAPI:

- H.323 for VoIP signalling
- H.450 for supplementary services
- T.38 for realtime fax

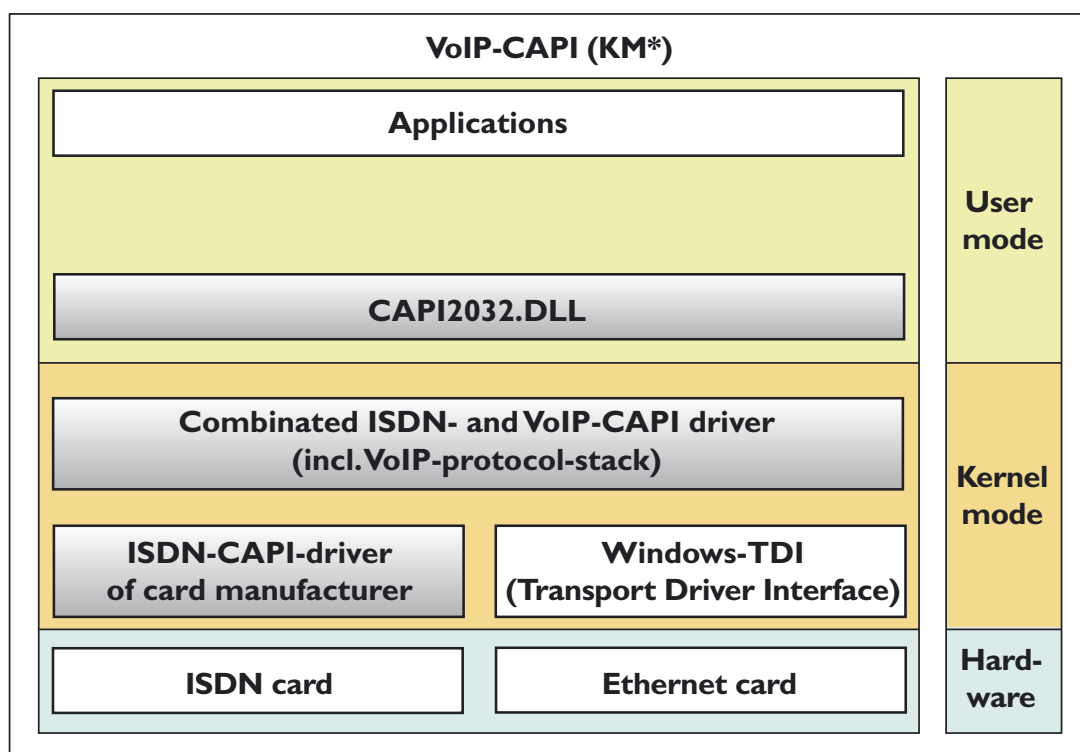
Other Functions

XCAPI operates a maximum of eight controllers, no matter whether they are for ISDN- or Ethernet-cards. Cards can also be mixed on a machine. On a PC XCAPI can control up to 240 lines. ISDN-cards must offer a CAPI 2.0 interface for kernelmode use. The number of lines available in the ISDN mode depends on the ammount of lines offered by the ISDN cards in use. It usually is 2 to 30 per card.

Technical Basis

The technical basis for XCAPI is a regular standard PC, running Windows 2000/NT. Also required is an Ethernet card, compatible to NDIS-5. If ISDN is needed, any ISDN card standardized to CAPI 2.0 can be used.

If there are any questions left, please do not hesitate to contact us. We'd like to give you further information about XCAPIs functions and its easy installation!



Studer OnAir Consoles



OnAir 3000 Modulo



OnAir 3000



OnAir 2000M2



OnAir 1000



OnAir 500 Modulo



OnAir 500

STUDER
professional audio equipment

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