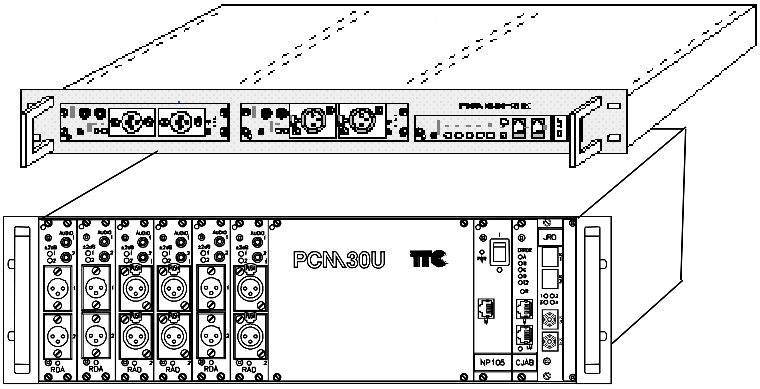

PCM30U-ROK

2 048/256 kbps

<u>Broadcast Codec</u>

Brief Overview

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Broadcast Codec PCM30U-ROK

A package of PCM30U broadcast codec equipment makes possible digital processing and transport of LF signal in analogue or digital form AES / EBU in the highest possible quality corresponding to the chosen transmission capacity. LF signal can be transmitted without compression, MPEG1-compressed, or possibly with the compression according to ITU-T J.41.

The package uses construction of 1U-height that is cost-effective and space-effective for max. 2 stereo channels, or a standard 3U PCM30U construction system for max. 7 stereo channels. In addition, the system allows for the transmission of voice and data signals – typically RDS (Radio Data System).

As a standard, the codec is equipped with 2 E1 interfaces. Optical 1st or 2nd order interfaces can be added, or it is possible to use redundant transmission over e.g. microwave radio.

The units carrying MPEG compression sub-units can be used also independently in V11 variant (without a central unit) to be directly connected to a standard 128 or 256 kbps, V.11 (RS422) channel.

The system is fed from 48 VDC. Mains power supply 100 ÷ 240 VAC can be delivered on demand.

RAD – Stereo LF Signal Encoder

The unit is determined for the transmission of one stereo or two monaural broadcast channels. LF signal is encoded in a selected mode after optional correction by pre-emphasis circuits according to ITU-T J.17.

There are located XLR5 connectors on the front panel for connecting input LF signal. Unit status is indicated by a bicolour LED.

It is possible to adjust input impedance, nominal input level, pre-emphasis, mono/stereo mode and HQ / MPEG / LIN18 / V11 operation by means of jumpers and switches of the unit.

RDA – Stereo LF Signal Decoder

The unit is determined for the transmission of one stereo or two monaural broadcast channels. The received signal is decoded in a selected mode and optionally corrected by de-emphasis circuits according to ITU-T J.17.

It is possible to adjust output impedance, nominal output level, type of compression, de-emphasis, mono/stereo mode and HQ / MPEG / LIN18 / V11 operation by means of jumpers and switches of the unit.

There are located XLR5 connectors on the front panel for connecting output LF signal. Unit status is indicated by a bicolour LED.

Unit Modes

- **LIN18(20Hz ÷ 21.7kHz)**

This mode offers the highest transmission quality. 18-bit linear transmission without compression occupies 28 time slots of 2 Mbps frame (1792 kbps).

- **MPEG (20Hz ÷ 20kHz)**

offers high-quality transmission at low demand for transmission capacity. On the transmitter side, the 18-bit signal is compressed in MPK sub-unit of MPEG1 encoder into 2 or 4 time slots (128 or 256 kbps). At the same time the high quality of transmitted signal is retained. The MPEG signal received on the receiver side is de-compressed in MPD sub-unit of decoder back to the 18-bit signal.

- **V.11**

Transmission properties are equal to those of MPEG.

- **HQ (20Hz ÷ 15kHz)**

14-bit compression (14/11 + 1 parity bit) with sample repetition at parity error according to ITU- J.41. Better parameters can be attained thanks to the modified compression using a free bit according to Rec. J.41 for the transmission of the 15th bit. The system conserves compatibility with 14-bit standard. In case of low error rate of 2Mbps signal, it is in place to use 16-bit proprietary TTC compression scheme (16/12 bits without a parity bit). The signal encoded this way occupies 12 time slots of 2 Mbps frame (768 kbps).

MPK - MPEG Encoder

RAD and AESR units can be completed with MPK sub-unit containing MPEG1 layer II compression encoder that makes it possible to increase the maximum transmitted LF frequency from 15 kHz to 20 kHz at decreasing the number of occupied 64 kbps channels. At preserving the high quality of the transmitted LF signal 20 Hz through 20 kHz, four 64 kbps channels are sufficient. Two 64 kbps channels are enough for 20 Hz through 11.8 kHz in „2 x mono“ mode (20 Hz through 13.4 kHz in „joint stereo“ mode) and LF transmission quality comparable with CD.

MPD – MPEG Decoder

RDA and AEST units can be completed with MPD sub-unit containing MPEG1, 2 layer II, III compression encoder that makes it possible to increase the maximum transmitted LF frequency from 15 kHz to 20 kHz at decreasing the number of occupied 64 kbps channels. At preserving the high quality of the transmitted LF signal 20 Hz through 20 kHz, four 64 kbps channels are sufficient. Two 64 kbps channels are enough for 20 Hz through 11.8 kHz and LF transmission quality comparable with CD. Error of MPEG decompression is indicated by the blinking of a red LED.

AESR – Receiver of AES / EBU Digital Signal

The unit receives digital music signal according to AES / EBU or SPDIF, extracts useful music signal out of it, as well as user and status bits, and gains clock signal. Music signal and user bits get processed for the transmission in 2 Mbps stream. The difference between transmission rates gets balanced by means of positive and negative stuffing. There are metallic and optical interfaces brought to the front panel.

AEST – Transmitter of AES / EBU Digital Signal

The unit receives digital music signal and user bits from 2 Mbps signal. In accordance with the received stuffing, it recovers the original clock frequency. Music signal gets transformed into AES / EBU form, and received user and status bits get inserted apart. The unit supports sampling frequencies 32 and 48 kHz. Sampling frequency 44.1 kHz is being prepared. There are metallic and optical interfaces brought to the front panel.

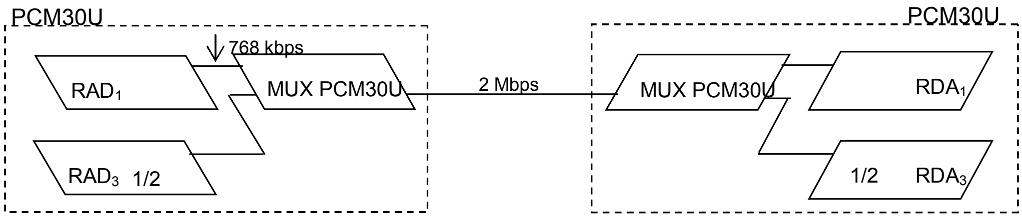
CJAB – Central Unit

contains two E1 interface circuits. The unit serves for transmission redundancy and for management over other units, and interfaces optical link terminal unit (JRO).

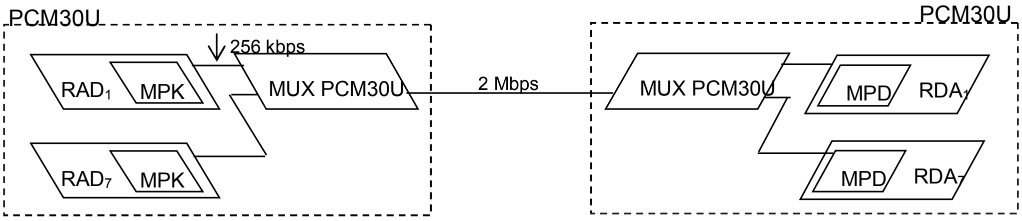
DV24 - V.24/V.28 (RS232) Interfaces

The unit carries two channel terminals of 64 kbps synchronous streams or two asynchronous streams up to 19.2 kbps. As a part of broadcast codec, it is suitable for the transmission of RDS signal.

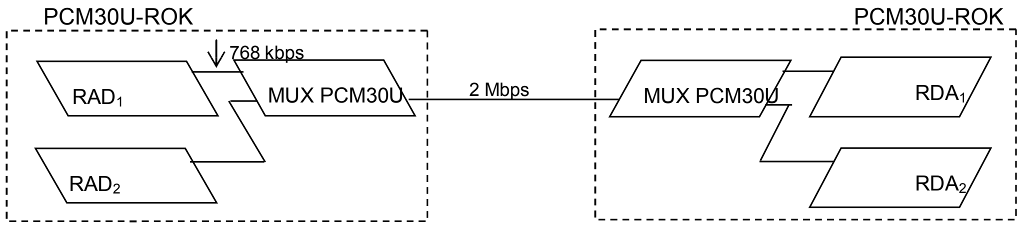
Examples of application of PCM30U broadcast signal transmission units and maximum number of LF channels that can be transmitted.



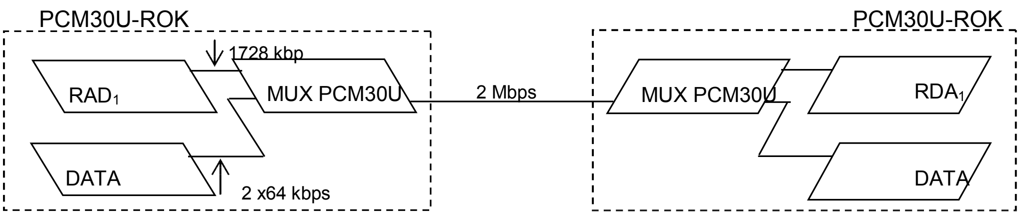
a) **LIN18: 1 x STEREO; HQ: 2 x STEREO + 1x MONO; MQ: 5 x STEREO**



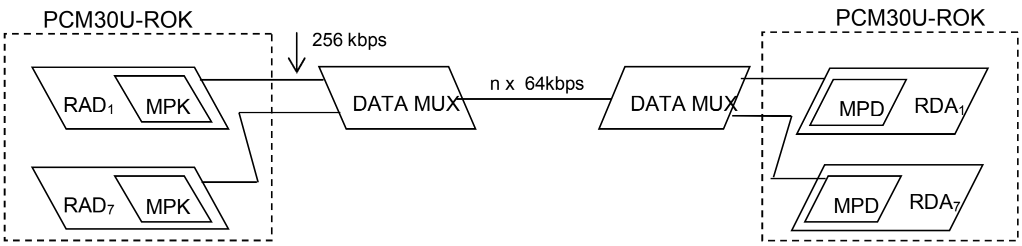
b) **MPEG256: 7x STEREO; MPEG128: 15 x STEREO**



c) **LIN18: 1 x STEREO; MPEG256/128, MQ, HQ: 2 x STEREO**



d) **LIN18: 1 x STEREO; DATA: 2 x 64 kbps**



e) **V11: 2 x STEREO**

Technical parameters of the variant without compression 18 bit lin 48kHz - LIN18

Transmitted bandwidth	20 Hz ÷ 21.7 kHz	
Sampling frequency	48 kHz	
Transmission rate stereo	1728 kbps	
	28 PCM 64 kbps channels	
Number of bits per sample	18, linear coding	
Codec delay	1.2 ms (+ delay of transmission path)	
Pre-emphasis / de-emphasis	ITU-T J.17 (attenuation 0 dB @ 2.1 kHz)	
Input signal level	- 3 dBm ÷ + 21 dBm	
Output signal level	- 3 dBm ÷ + 21 dBm for R_{out} < 20Ω	
	- 9 dBm ÷ + 15 dBm for R_{out} = R_{load} = 600Ω	
Input impedance – R_{in}	> 15 kΩ or 600Ω (balanced)	
Output impedance – R_{out}	< 20Ω or 600Ω (balanced)	
Inaccuracy of residual attenuation adjustment @ 0 dBr (1 kHz)	< ± 0,2 dB (± 2 dB on demand)	
Level of limiting	15 dBm0 (may be 12 dBm0)	
Common mode rejection ratio (CMRR)	≥ 70 dB	
Channel attenuation distortion		
20 Hz ÷ 21.5 kHz	< ± 0.1 dB	
21.5 kHz ÷ 22 kHz	< + 0 / - 1.4 dB	
> 26.4 kHz	< - 85 dB	
Group delay variation	20 Hz ÷ 21.7 kHz	< 0.02 ms
Dynamic range (20 Hz ÷ 21.7 kHz)	> + 98 dB (typ. 102 dB)	
Total harmonic distortion + noise		
-1 dB	< - 89 dB (typ 93 dB)	
-20 dB	< - 78 dB (typ 82 dB)	
60 dB	< - 38 dB (typ 42 dB)	
Crosstalk between channels at selective measurement by sinusoidal signal @ 0 dBm0	20 Hz ÷ 21.7 kHz	> + 85 dB
Level difference between L and R channels	20 Hz ÷ 21.7 kHz	< ± 0.25 dB
Phase difference of L and R channels	20 Hz ÷ 21.7 kHz	< ± 0.8°
Output level of audio monitoring	0 ± 0.2 dBm @ level of limiting	R_{load} = R_{out} = 600Ω (unbal.)

Note: all parameters without pre-emphasis / de-emphasis.

Naformátováno: Angličtina (USA)

Technical parameters of variant with MPEG 1 LII - MPEG256, MPEG128 compression

Channel type	**MPEG256**	**MPEG128**
Transmitted bandwidth - joint stereo	20 Hz ÷20 kHz	20 Hz ÷ 13.2 kHz
- 2 channels	20 Hz ÷20 kHz	20 Hz ÷ 11.8 kHz
Sampling frequency	48 kHz	32 kHz
Transmission rate stereo	256 kbps	128 kbps
	4 PCM 64 kbps channels	2 PCM 64kbps channels
Codec delay (without transmission path)	100 ms	150 ms
Number of bits per sample	20 (AD converter 20bit, DA converter 24bit)	
Channel attenuation distortion		
256@48	20 Hz ÷ 20 kHz	< ± 0.1 dB
	> 20.6 kHz	< - 85 dB
128@32 (joint stereo)	20 Hz ÷ 13.2 kHz	< ± 0.1 dB
	> 13.8 kHz	< - 85 dB
Compression response	MPEG1 Layer II:	- stereo - joint stereo - 2 channels - 1 channel *)
Pre-emphasis / de-emphasis	ITU-T J.17 (attenuation 0 dB @ 2.1 kHz)	
Input signal level	- 3 dBm ÷ + 21 dBm	
Output signal level	- 3 dBm ÷ + 21 dBm pro R_{out} < 20Ω	
	- 9 dBm ÷ + 15 dBm pro R_{out} = R_{load} = 600Ω	
Input impedance – R_{in}	> 15 kΩ or 600Ω (balanced)	
Output impedance - R_{out}	< 20Ω or 600Ω (balanced)	
Inaccuracy of residual attenuation adjustment 0 dBr (1 kHz)	< ± 0,2 dB (± 2 dB on demand)	
Level of limiting	15 dBm0 (may be 12 dBm0)	
Common mode rejection ratio (CMRR)	≥ 70 dB	
Group delay variation	20 Hz ÷ 20 kHz (13.2 kHz)	< 0.02 ms
Dynamic range (20 Hz ÷ 20 kHz)	> + 96 dB (typ. 98 dB)	> + 94 dB (typ. 96 dB)
Total harmonic distortion + noise - 1 dB - 20 dB - 60 dB	 < -89 dB (typ. 91 dB) < - 78 dB (typ. 80 dB) < - 38 dB (typ. 40 dB)	 < - 89 dB (typ. 91 dB) < - 74 dB (typ. 76 dB) < - 34 dB (typ. 36 dB)
Crosstalk between channels at selective measurement by sinusoidal signal @ 0 dBm0	20 Hz ÷ 20 kHz (13.2 kHz)	> + 85 dB
Level difference between L and R channels	20 Hz ÷ 20 kHz (13.2 kHz)	< ± 0.25 dB
Phase difference of L and R channels	20 Hz ÷ 20 kHz (13.2 kHz)	< ± 0.8°
Output level of audio monitoring	0 ± 0.2 dBm at level of limiting	R_{load} = R_{out} = 600Ω (unbal.)

Note: all parameters without pre-emphasis / de-emphasis.
*) On demand, program memory can be replaced.

Technical parameters of variant with compression ITU-T J.41 (J.42) - HQ, MQ

Transmitted bandwidth	**HQ**	**MQ**
	20 Hz ÷ 15 kHz	20 Hz ÷ 7.5 kHz
Sampling frequency	32 kHz	16 kHz
Transmission rate - mono	384 kbps	192 kbps
	6 PCM 64 kbps channels	3 PCM 64 kbps channels
Transmission rate - stereo	768 kbps	384 kbps
	12 PCM 64 kbps channels	6 PCM 64 kbps channels
Number of bits per sample	- standard compression J.41 (HQ), J.42 (MQ) 14 - modified compression J.41 (HQ), J.42 (MQ) 15 - compression TTC (both HQ and MQ) 16	
Compression response	- standard ITUT J.41 (J.42) 11-section response with instantaneous compression 14/11 + 1 parity bit with sample repetition at parity error - modified ITU-T J.41 (J.42) 13-section response with instantaneous compression 15/11 + 1 parity bit with sample repetition at parity error - TTC proprietary 13-section response with instantaneous compression 16 / 12 bits without parity bit	
Pre-emphasis / de-emphasis	ITU-T J.17 (attenuation 0 dB @ 2.1 kHz)	
Input signal level	- 3 dBm ÷ + 21 dBm	
Output signal level	- 3 dBm ÷ + 21 dBm pro R_{out} < 20Ω	
	- 9 dBm ÷ + 15 dBm pro R_{out} = R_{load} = 600Ω	
Input impedance - R_{in}	> 15 kΩ or 600Ω (balanced)	
Output impedance - R_{out}	< 20Ω or 600Ω (balanced)	
Inaccuracy of residual attenuation adjustment @ 0 dBr (1 kHz)	< ± 0,2 dB (± 2 dB on demand)	
Level of limiting	15 dBm0 (may be 12 dBm0)	
Common mode rejection ratio (CMRR)	≥ 70 dB	
Channel attenuation distortion		
HQ	20 Hz ÷ 14.4 kHz	< ± 0.1 dB
	14.4 kHz ÷ 15 kHz	< + 0 / - 1.4 dB
	> 17.6 kHz	< - 85 dB
MQ	20 Hz ÷ 7.2 kHz	< ± 0.1 dB
	7.2 kHz ÷ 7.5 kHz	< + 0 / - 1.4 dB
	> 8.8 kHz	< - 85 dB
Group delay variation	20 Hz ÷ 14.4 kHz (7.2 kHz)	< 0.02 ms
S/N ratio of sinusoidal signal - 20 through +15 dBm0 @ f = 1 kHz / total distortion and noise	- standard compression J.41 (J.42) - modified compression J.41 (J.42) - compression TTC	> + 50 dB (dist.< 0.32%) > + 56 dB (dist.< 0.16%) > + 62 dB (dist.< 0.08%)
Idle channel noise (not weighted – effective value)	- standard compression J.41 (J.42) - modified compressionJ.41 (J.42) - compression TTC	< - 64 dBm0 < - 70 dBm0 < - 75 dBm0
Crosstalk between channels at selective measurement by sinusoidal signal @ 0 dBm0	20 Hz ÷ 14.3 kHz (7.1 kHz)	> + 80 dB
Level difference between L and R channels	20 Hz ÷ 14.3 kHz (7.1 kHz)	< ± 0.25 dB
Phase difference of L and R channels	20 Hz ÷ 14.3 kHz (7.1 kHz)	< ± 0.8°
Output level of audio monitoring	0 ± 0.2 dBm @ level of limiting	R_{load} = R_{out} = 600Ω (unbal.)

Note: all parameters without pre-emphasis / de-emphasis.