Introduction to DAB Receiver

尤信程
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Contents

- DAB broadcasting in Taiwan
- ETSI 300 401 (Eureka-147) standard
- Analog/Digital front end
- OFDM demodulation
- Time/Frequency synchronization
- Channel decode
- Conclusions

DAB broadcasting in Taiwan (1)

時間	政府 單位	發展記事
1995	經濟部	委託工研院電通所執行『數位廣播系統技術』專科計劃。
1998	交通部電信總局	擬定DAB推動草案。
1999	新聞局	召開『草擬廣電法數位廣播電視相關條文會議』，並建議採Eureka-147及VHF Band III試播。
2000	交通部電信總局	核定台灣全區2個、北區3個、中區2個、南區3個系統進行試播，並陸續開播。
2005	新聞局	公告第一梯次數位廣播審議結果。

DAB broadcasting in Taiwan (2)

播放區域	早期試播電台名稱	發射頻率
全區	中廣	220.352MHz
	中央(警廣、漢聲、教育)	222.064MHz
單區	台北之音(1)、 人人(1)、高屏(3)	211.648MHz
	亞洲(1)、全國(2)、大眾(3)	213.360MHz
	飛碟(1)、真善美(2)、南台灣(3)	215.072MHz

DAB broadcasting in Taiwan (3)

- 2005.06.26: 新聞局公告第一梯次數位廣播審議結果
- 全區: 福爾摩沙(民視) 、優越傳信(高雄大眾)、中廣
- 北區: 寶島新聲、台倚(台哥大，倚天， 台北之音，新竹IC之音)
- 中區: 從缺
- 南區: 好事數位(高雄港都)
- 尚有中區兩張及南區一張執照未核定，待第二次申請。

DAB broadcasting in Taiwan (2)

播放區域	正式開播電台名稱	發射頻率
全區	福爾摩沙	11 D (222.064)
	優越傳信	11 C (220.352)
	中廣	10 D (215.072)
單區	寶島新聲 (1)	10 C (213.360)
	台倚 (1)	10 B (211.648)
	好事數位 (3)	10 C (213.360)

DAB broadcasting in Taiwan (4)

- 數位廣播開放原則
  - 音訊50 %以上，其餘可為數據廣播
  - 最少一個免費頻道(audio 192 kbps or speech 96 kbps)
  - 加強防災能力
- 時程
  - 六個月內申請電台許可
  - 三年內30 %涵蓋率
  - 九年內全區營運

DAB broadcasting in Taiwan (5)

- 交通部未來可能開放美規及短波數位廣播
- 美規數位廣播(HD radio or IBOC)
  - 與FM共存於相同頻率範圍(Simulcast)
  - 美加以外，未有其他國家採用
- 短波數位廣播(DRM, Digital Radio Mondale)
  - 使用短波(short wave)
  - 訊號頻寬9 kHz
  - 調變方法接近DAB
  - 使用HE-AAC
  - 歐洲有許多電台開始廣播

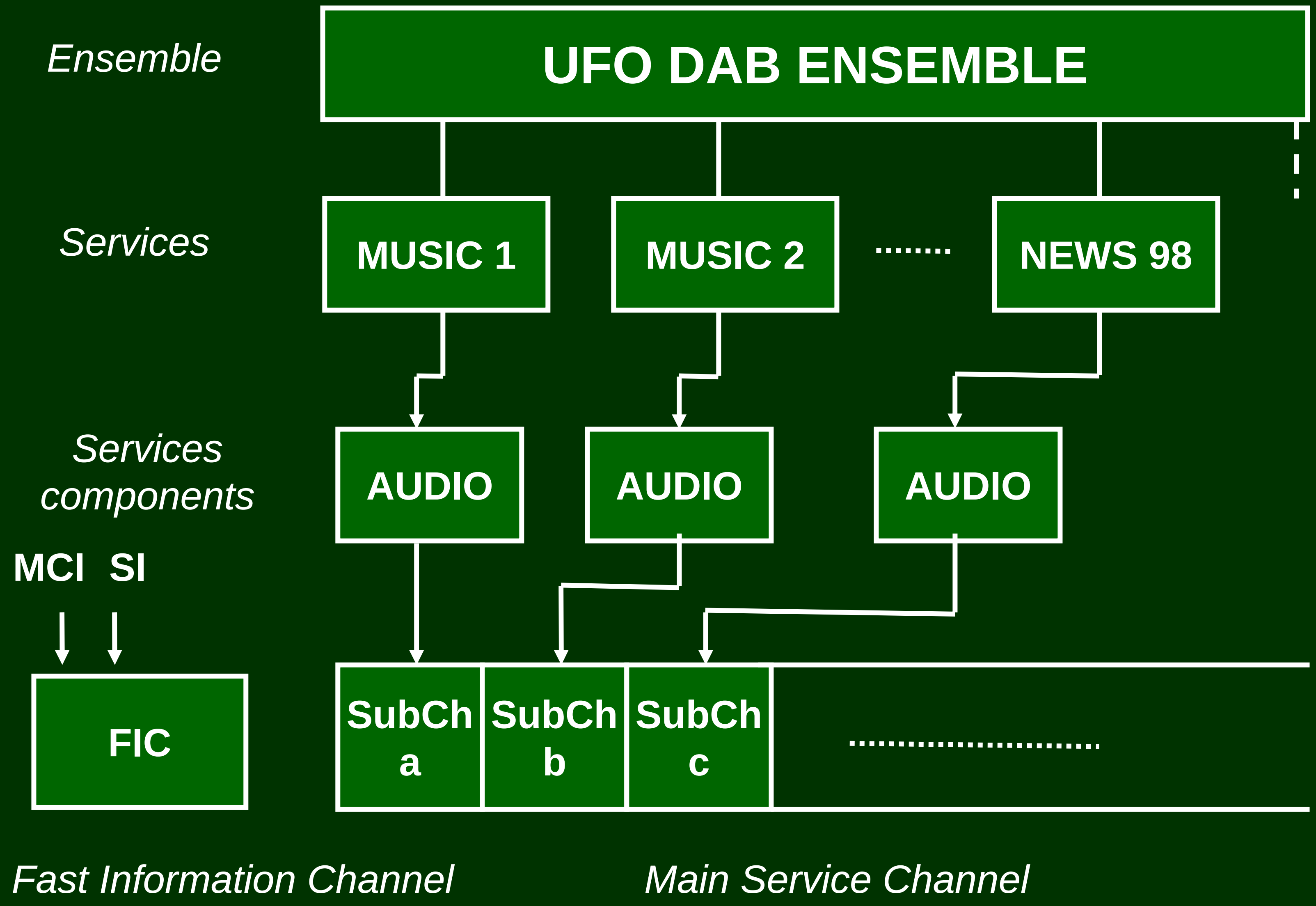
ETSI 300 401 standard (1)

- Also known as Eureka-147 system
- Developed jointly by EBU, CENELEC, and ETSI
- Edition-1 standard available in 1995
- Mainly for terrestrial broadcasting (Band I-V, L-band). It can also be used for satellite broadcasting
- Widely used in lots of areas, including Taiwan and China

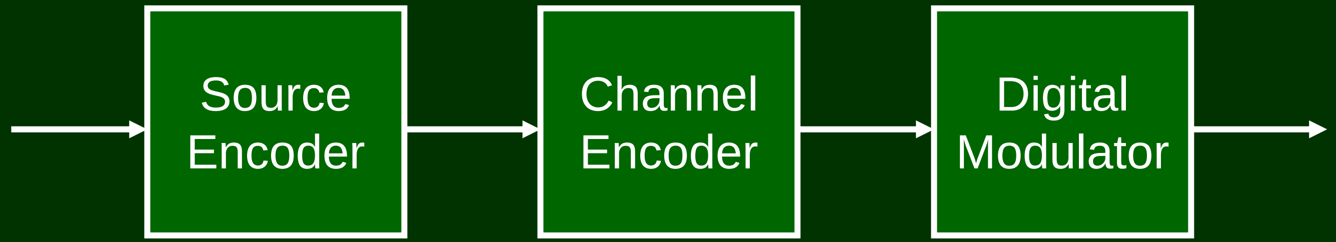
ETSI 300 401 standard (2)

- Four different modes available with different FFT lengths: 2048, 512, 256, and 1024
- Transmission bandwidth around 1.5 MHz
- Multiple audio and data services in an ensemble
- Streaming mode and packet mode for data transmission
- Many stations may construct a single frequency network (SFN)

ETSI 300 401 standard (3)



ETSI 300 401 standard (4)

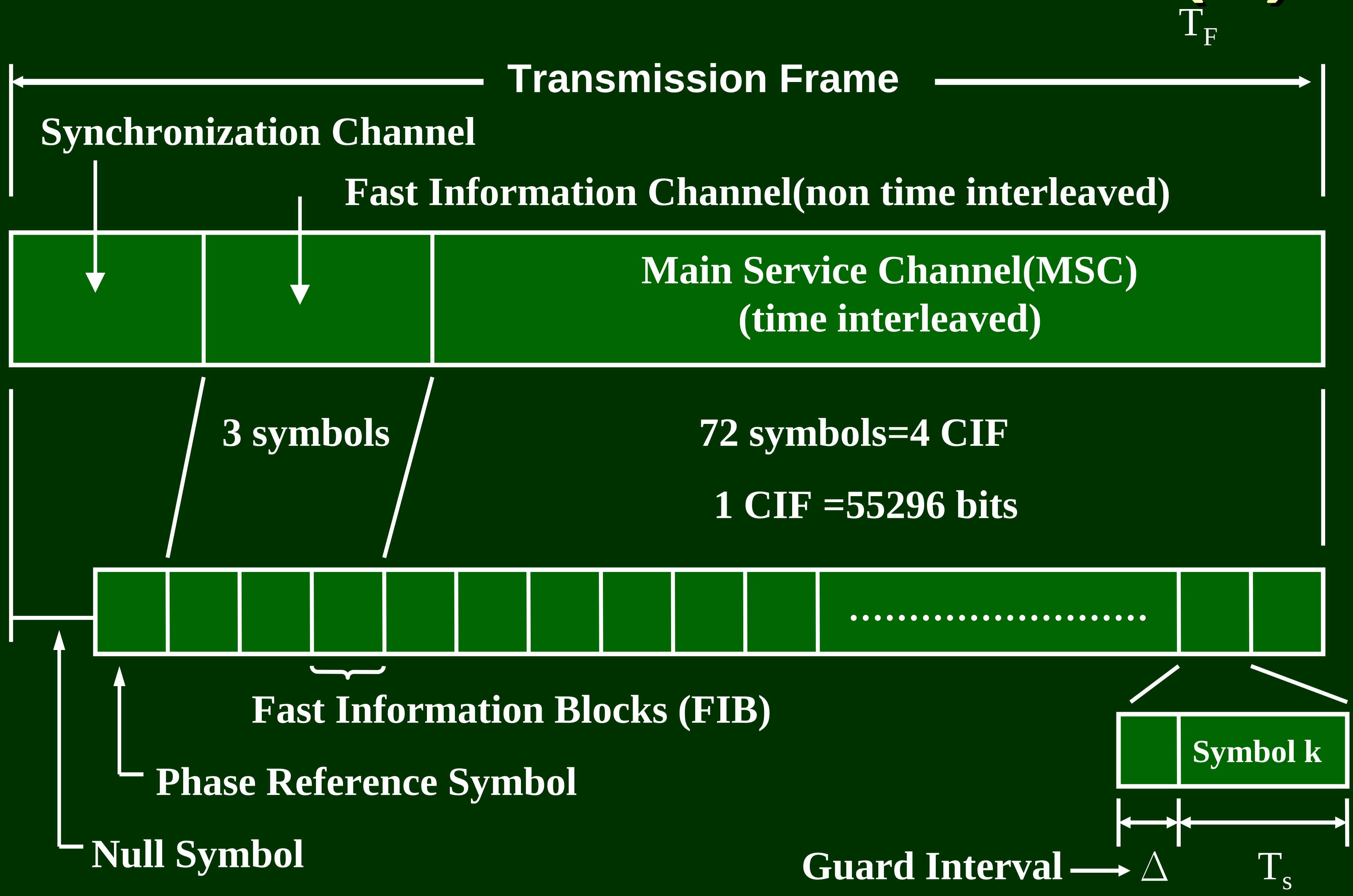


- Source encoder: MPEG 1/2 Layer II
- Channel encoder: energy dispersal, convolutional encode, time interleaving
- Digital modulator: ¼-π DQPSK, frequency interleaving, OFDM

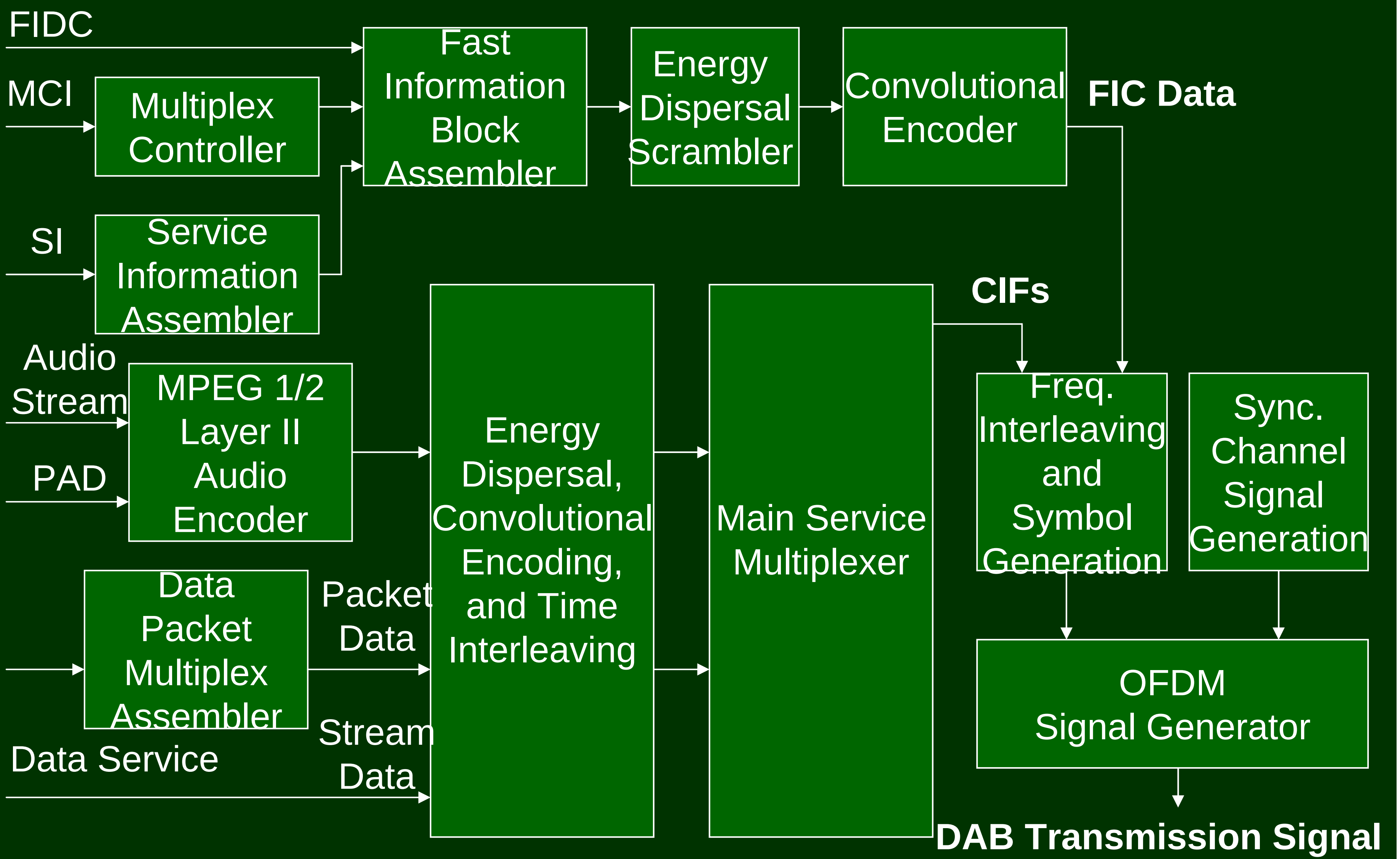
ETSI 300 401 standard (5)

- Transmission Frame (TF): Sync Channel (SC), Fast Info Channel (FIC), and Main Service Channel (MSC)
- Mode I example (used in Taiwan): TF = 96 ms, SC ~ 2.5 ms, FIC ~ 3.7 ms, MSC ~ 89.7 ms
- Mode I example: FIC = 12 FIB (Fast Info Block) MSC = 4 CIF (Common Interleaved Frame)
- One CIF = 24 ms is a logical frame

ETSI 300 401 standard (6)



ETSI 300 401 standard (7)



Introduction to SFN (1)

- Conventional TV/Radio systems cannot be used to build a SFN (Single Frequency Network).
- There are Interferences between different transmitters (stations) of the same TV/Radio company.
- We need to change the frequency of the radio in order to keep listening to the traffic info in a highway.

Introduction to SFN (2)

- Since multiple carriers with guard interval can deal with the ISI, the same tech is also used to overcome the interference due to different transmitters.
- Therefore, with proper design, OFDM tech may be used to build the SFN.

Introduction to SFN (3)

- In a SFN, the max distance between two transmitters is determined by the guard interval.
- Precise time sync between transmitters is implemented by using the signal from GPS.
- Currently, TV/Radio companies have constructed their SFN’s for digital TV and digital radio.

Introduction to SFN (4)

- With the SFN of a radio station, we don’t need to change the dial of the radio from Taipei to Kaohsiung.
- Constructing SFN’s actually improve the spectrum efficiency. For example, ICRT uses 100.7 in Taipei and 100.1 in central Taiwan. With SFN tech, ICRT needs only 100.7 and 100.1 can be allocated to some one else.

OFDM generation (1)

- OFDM signal *s*(*t*) can be generated by using many isolated carriers. In fact, this is not necessary if "digital tech" is used.
- Recall $s(t) = \sum z_m e^{jm\omega_0 t}$

  If we use the identity $T_0 = 2\pi / \omega_0$

  then,

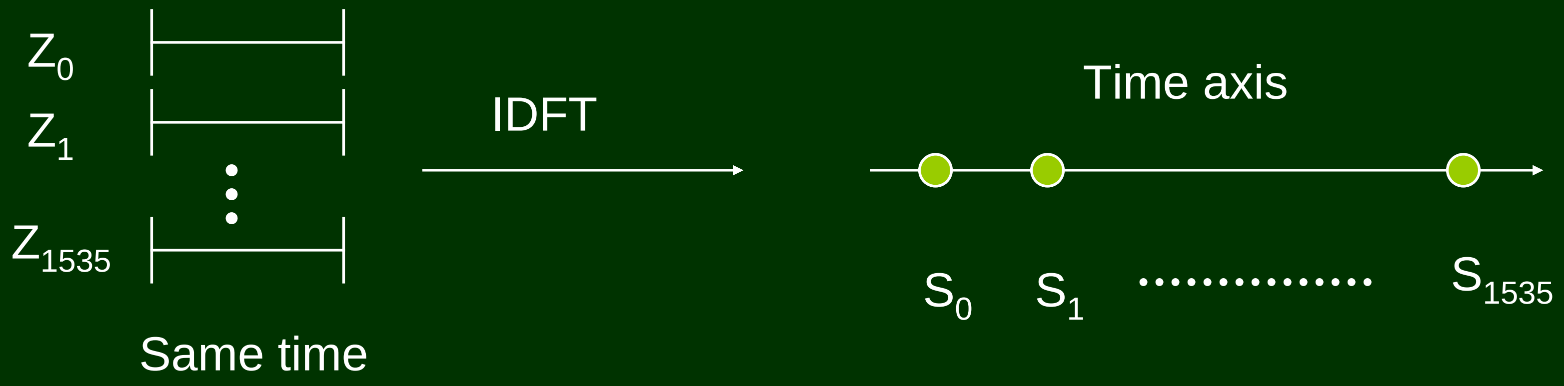
$$s(t) = \sum_{m=0}^{M-1} z_m e^{j\frac{2\pi}{T_0}mt}$$

OFDM generation (2)

- If the data is sampled by *N* points in the T_0 interval, then the equation becomes

$$s(n\cdot\Delta t)=\sum_{m=0}^{M-1} z_m e^{j\frac{2\pi}{N\cdot\Delta t}mn\cdot\Delta t}=\sum_{m=0}^{M-1} z_m e^{j\frac{2\pi}{N}mn}$$

This is similar to the IDFT equation.

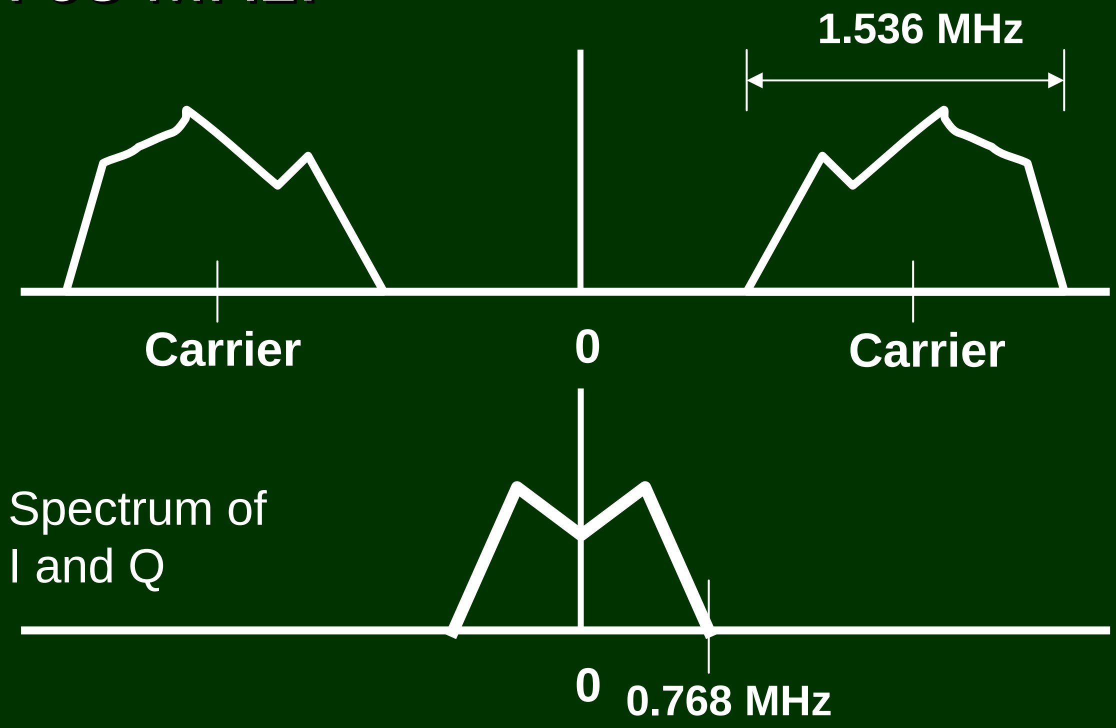


OFDM generation (3)

- In DAB mode I, M = 1536, N = 2048. Why M is not equal to N?
- Sampling theorem: sampling rate >= 2 BW (base band)
- To allow for some safe margin, we usually use a sampling rate of more than 2 BW.
- With N = 2048 and $\omega_0 / 2\pi$ = 1 kHz, f_s = 2.048 MHz. Thus, BW in base band < 1.024 MHz.

OFDM generation (4)

- The OFDM signal has a BW of 1.536 MHz. It's composed of I and Q in base band with BW = 0.768 MHz.



OFDM generation (5)

- To be efficient, we let $M = N$ and $z_m = 0$ for $(M - 1) < m < N$
- We then can use IFFT to implement the OFDM generation. (Use k to replace m)

$$s(n) = \sum_{k=0}^{N-1} z_k e^{j\frac{2\pi}{N}kn}$$

Format of transmit signal :

$$s_m(t) = \mathrm{Re}\{e^{2j\pi\pi_c t} \cdot s(t)\}$$

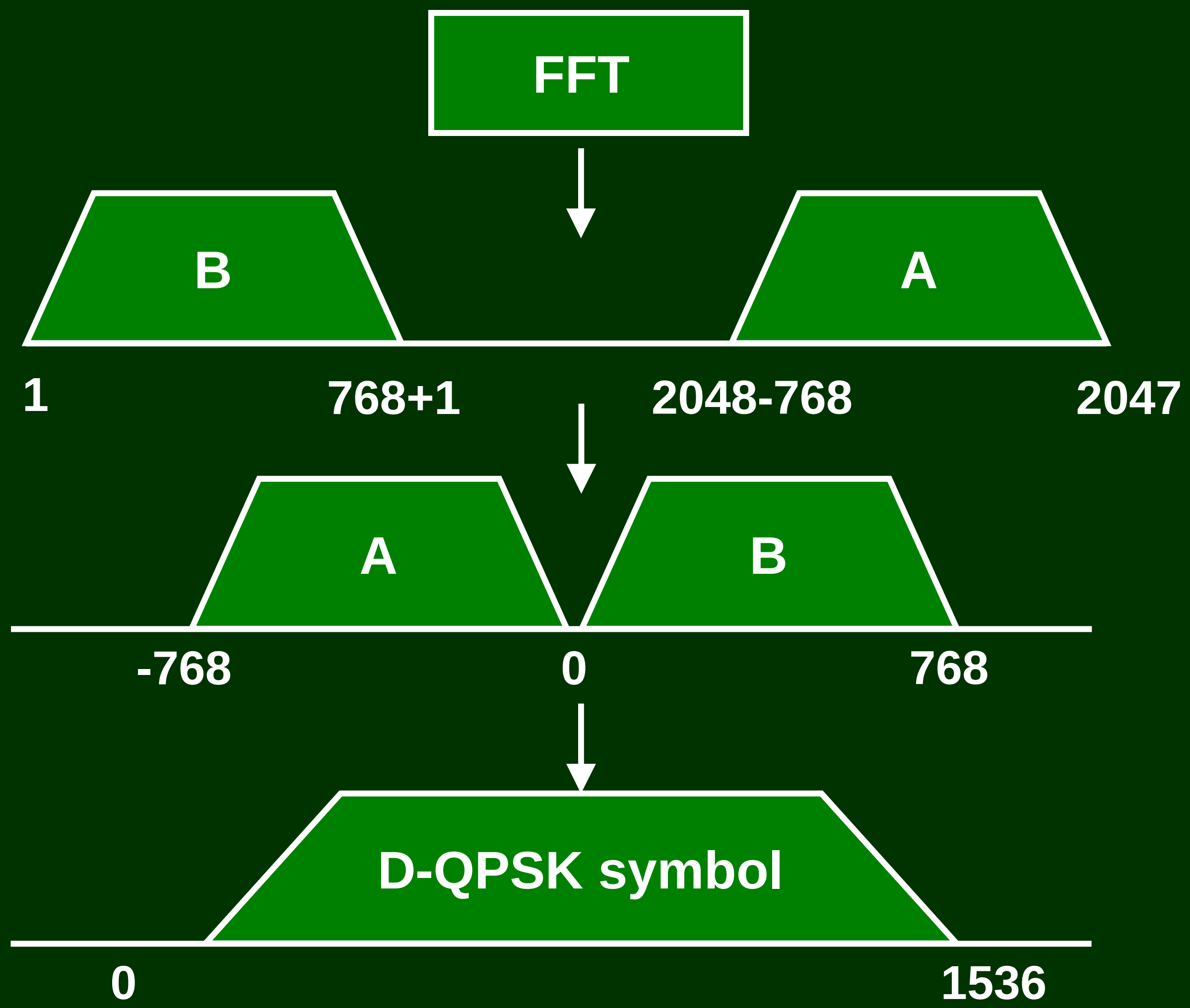
OFDM generation (6)

- The same ideal applies to OFDM demodulation.
- We may use summation to replace integral. Thus, the equation becomes DFT. Since we use *M* = *N*, care must be taken to deal with this.

$$\int s(t)e^{-jk\omega_0 t}dt \Rightarrow \sum_{n=0}^{M-1} s(n)e^{-j\frac{2\pi}{N}kn}$$

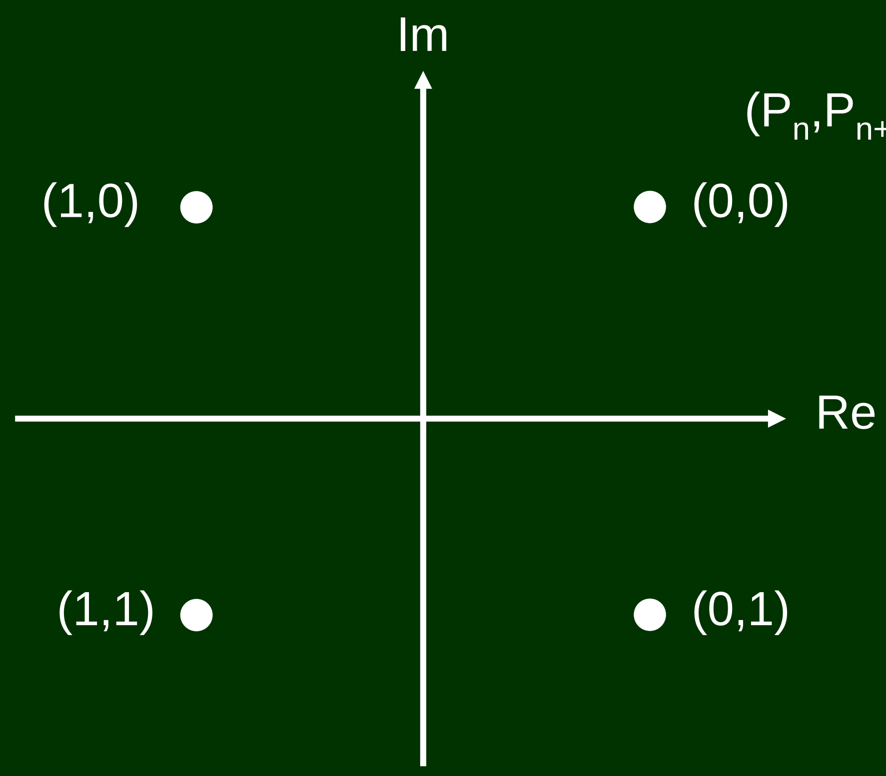
$$\Rightarrow \sum_{n=0}^{N-1} s(n)e^{-j\frac{2\pi}{N}kn}$$

OFDM generation (7)



π /4-DQPSK (1)

$q_n = 1/\sqrt{2}[(1-2p_n)+j(1-2p_{n+1536})]$

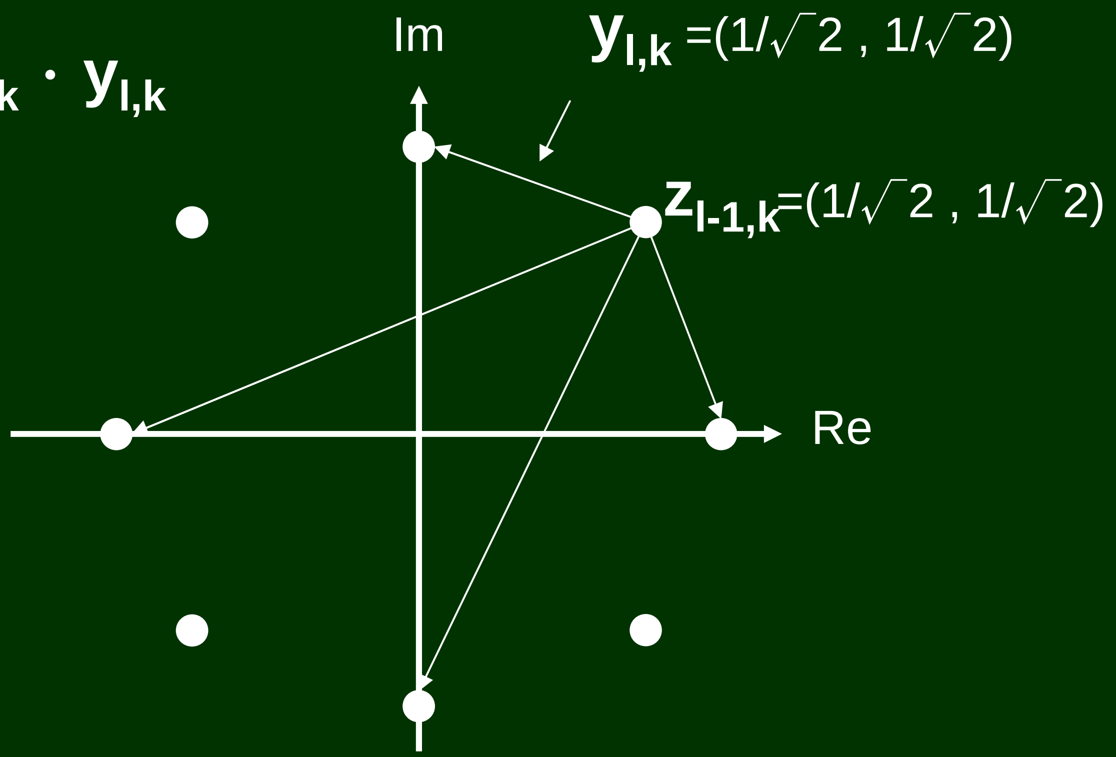


3072 vector->1536 QPSK symbol

π/4-DQPSK (2)

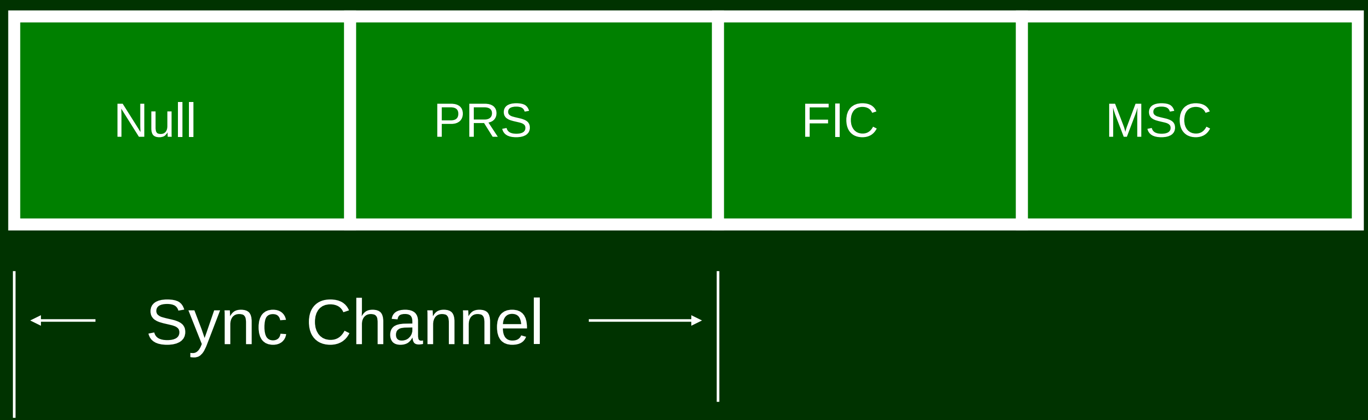
Using differential modulation relaxes the time synchronization requirements.

$z_{l,k} = z_{l-1,k} \cdot y_{l,k}$



Sync channel (1)

- Sync channel contains null symbol and Phase Reference Symbol (PRS).
- Null symbol has a much lower energy. It contains nothing but TII (Transmitter Identification Information)



Sync channel (2)

- PRS has 4 blocks with each block having 32 values. The allowed values are {+1,-1,+j,-j}.
- For DAB in mode 1, there are 1536 carriers.

  The carriers are divided into 12 groups. Each group has a constant bias (i.e., *n*).
- The equation of $z_{1,k}$ (i.e., PRS) is

$$z_{1,k} = \begin{cases} e^{j\varphi_k}, -768 \le k < 0 \ \text{ and } \ 0 < k \le 768 \\ 0, k = 0 \end{cases}$$

$$\varphi_k = \frac{\pi}{2}\left(h_{i,k-k'} + n\right)$$

Sync channel (3)

- There is a table providing the relationship between k and k', i, and n. (Table 44 – 47)
- There is another table for $h_{i,j.}$ (Table 48)
- Example of Table 44 (first 2 entries)

  $-768 \leq k \leq -737 \Rightarrow k' = -768, i = 0, n = 1$

  $-736 \leq k \leq -705 \Rightarrow k' = -736, i = 1, n = 2$
- Example of Table 48

  $h_{0,0} = 0, \quad h_{0,1} = 2, \quad h_{0,2} = 0, \ldots$

  $h_{1,0} = 0, \quad h_{1,1} = 3, \quad h_{1,2} = 2, \ldots$

Sync channel (4)

- Define the autocorrelation function $R(m)$ of $z(n)$ as $(-8 \le m \le 8)$

$$R(m) = \sum_{n=0}^{15} z(n) z^* < (n+m) >_{16}$$

- Let $R_1(m)$ is for $z_{1,k}$ with $-768 \le k \le -753$
  Let $R_2(m)$ is for $z_{1,k}$ with $-736 \le k \le -721$
  Let $R_3(m)$ is for $z_{1,k}$ with $-704 \le k \le -689$
  Let $R_4(m)$ is for $z_{1,k}$ with $-672 \le k \le -657$

Sync channel (5)

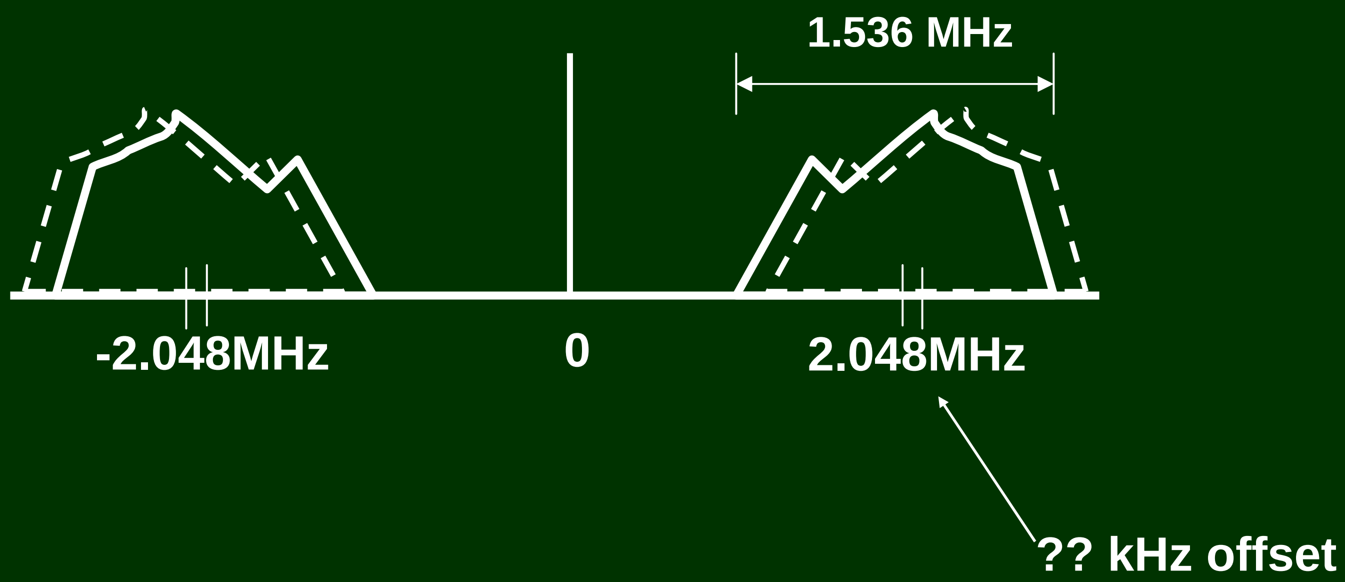
- The summation of $R_1(m)$ to $R_4(m)$ is an impulse with the only peak at $m = 0$.
- This can be used as a reference signal for receivers. The receiver must have a copy of the PRS inside it.
- To use PRS, time sync is **required** although the DQPSK does not!
- Usually we use PRS for coarse freq sync.

OFDM synchronization (1)

- OFDM needs to synchronize the followings
  - Receiver frequency
  - Symbol position (time)
  - Sampling rate

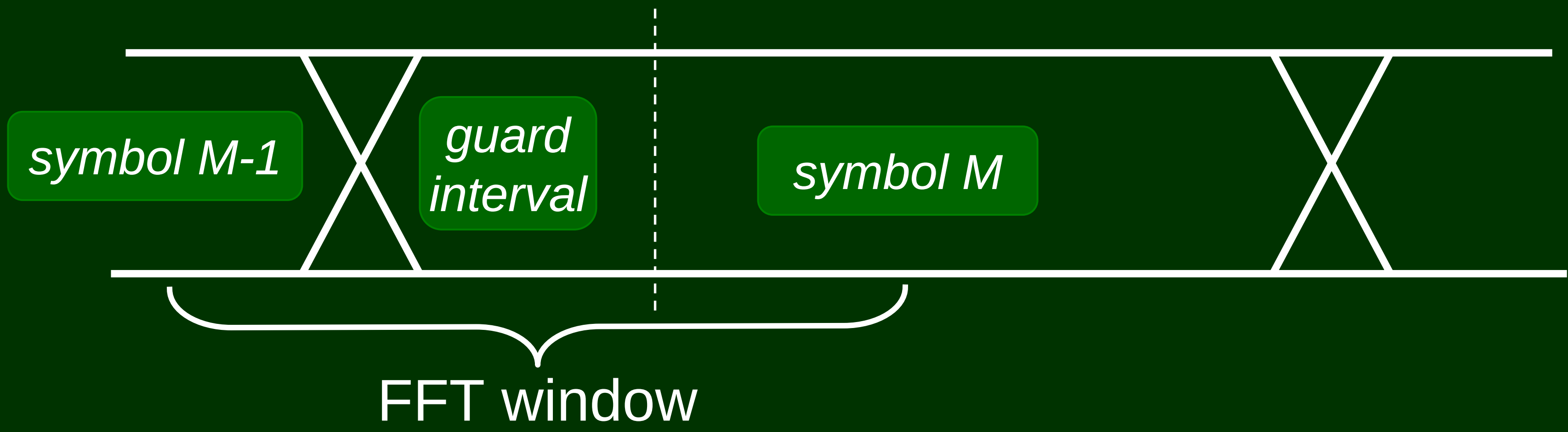
OFDM synchronization (2)

- Local Oscillator offset (Freq. Sync.)



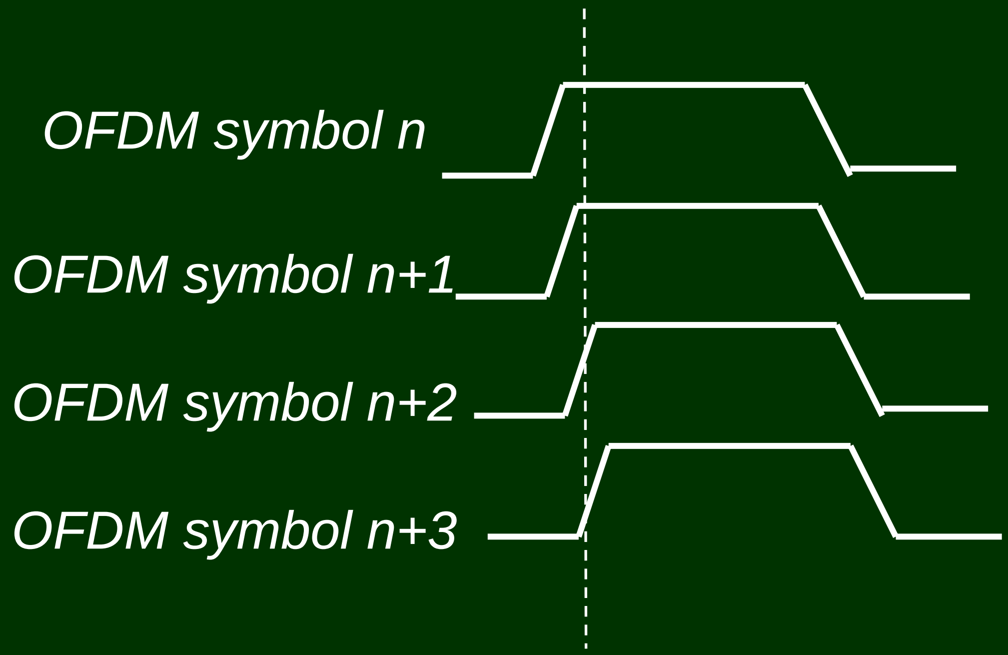
OFDM synchronization (3)

- Time sync to avoid incorrect position of FFT window.



OFDM synchronization (4)

- Sampling rate synchronization.



Frequency sync (1)

- Recall that OFDM requires freq synch
- Frequency synchronization can be divided into two classes:
  - Coarse freq sync (integer multiple of ω_0)
  - Fine freq sync (less than 0.5 ω_0)
- Incorrect coarse freq sync causes incorrect indexing of QPSK symbols
- Incorrect fine freq sync causes Inter-Carrier Interference (ICI)

Frequency sync (2)

- Consider coarse freq sync problem. Want to get z_k but receiver freq is $(k+k_0)\,\omega_0$ where k_0 is an integer. Thus,

$$z'_k = \sum_{n=0}^{N-1} s(n) e^{-j\frac{2\pi}{N}n(k+k_0)}$$

$$= \sum_{n=0}^{N-1} \sum_{m=0}^{N-1} z_m e^{j\frac{2\pi}{N}nm} e^{-j\frac{2\pi}{N}n(k+k_0)}$$

$$= N \cdot z_{(k+k_0)}$$

Frequency sync (3)

- Previous equation says that QPSK symbols are not correctly indexed. Need to re-assign.
- One way to find the coarse freq offset is via PRS. If position of OFDM symbol in TF (Transmit Frame) is **correct (usually not the case)**, then offset can be estimated via performing correlation between received PRS and build-in PRS.

Frequency sync (4)

- Influence of fine frequency offset. To recover z_k, but use $(k+\alpha)\,\omega_0$ where $\alpha \in [-0.5, 0.5]$. The result becomes

$$z'_k = \sum_{n=0}^{N-1} s(n) e^{-j\frac{2\pi}{N}n(k+\alpha)}$$

$$= \sum_{n=0}^{N-1} \sum_{m=0}^{N-1} z_m e^{j\frac{2\pi}{N}nm} e^{-j\frac{2\pi}{N}n(k+\alpha)}$$

$$= \sum_{m=0}^{N-1} z_m \frac{1-e^{j2\pi(m-k-\alpha)}}{1-e^{j(2\pi/N)(m-k-\alpha)}}$$

Frequency sync (5)

- The second term of the above equation is like a sinc function. It says that z_k' is contaminated by other QPSK symbols (ICI).
- Usually 2 % of inaccuracy is allowed. DAB mode I uses $2\pi\ \omega_0$ = 1 kHz. So, 20 Hz offset is allowed.

Frequency sync (6)

- One simple way to find the fine offset of frequency is by “searching.” A brute-force method. It is robust, but slow.
- Other methods available. EX: By using correlation between guard interval and corresponding part of OFDM symbol. (cf. IEEE trans SP, vol. 45, no. 7, pp 1800-1805, 1997)

Frequency sync (7)

- If α is obtained, then it is easy to cancel the offset by multiplying *s*(*n*) by $e^{j(2\pi/N)n\alpha}$ before FFT.

$$z'_k = \sum_{n=0}^{N-1} s(n) e^{j\frac{2\pi}{N}n\alpha} \cdot e^{-j\frac{2\pi}{N}n(k+\alpha)}$$

$$= \sum_{n=0}^{N-1} s(n) \cdot e^{-j\frac{2\pi}{N}nk}$$

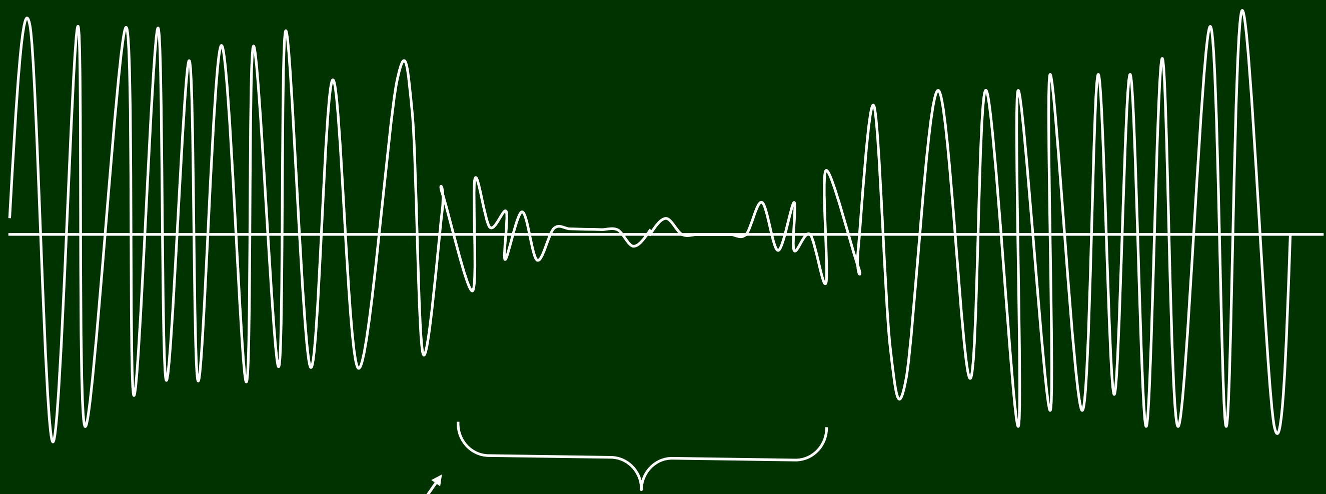
$$= z_k$$

Time synchronization (1)

- Exact time sync is not required for DQPSK. But it is needed to use PRS and to perform sampling-rate sync.
- Coarse time sync (in TF) is done via null symbol.
- Fine time sync is done via correlation in the time domain (i.e., before FFT).

Time synchronization (2)

- Detection of null symbol via envelope energy.



About 30-sample error

$$\frac{1}{2656}\sum_{n=0}^{2656} r_n{}^2$$

Average Energy

Time synchronization (3)

- Why time synchronization is a must for PRS? Let $s[<n-n_0>] = s[N-n_0],..s[N-1],s[0],.\ s[N-n_0+1]$, instead of $s[0],...s[N-1]$, be feed to FFT for OFDM demod. Then, FFT result is phase shifted.
- $\mathrm{FFT}(x[n-n_0]) \Rightarrow e^{-j(2\pi k/N)n_0} X(k)$
- It is obvious that time sync is required, or correlation won't work!

Time synchronization (4)

- Suppose the build-in PRS is called z_k.
- The received PRS (after FFT) is x_k.
- Correlating both signals and then taking IFFT, yields an impulse if no frequency offset. Use it to find time offset.

Time synchronization (5)

- If x_k is a time-shifted version of z_k before FFT, then $x_k = z_k\, e^{-j(2\pi k/N)n_0}$
- Recall assuming no freq offset.

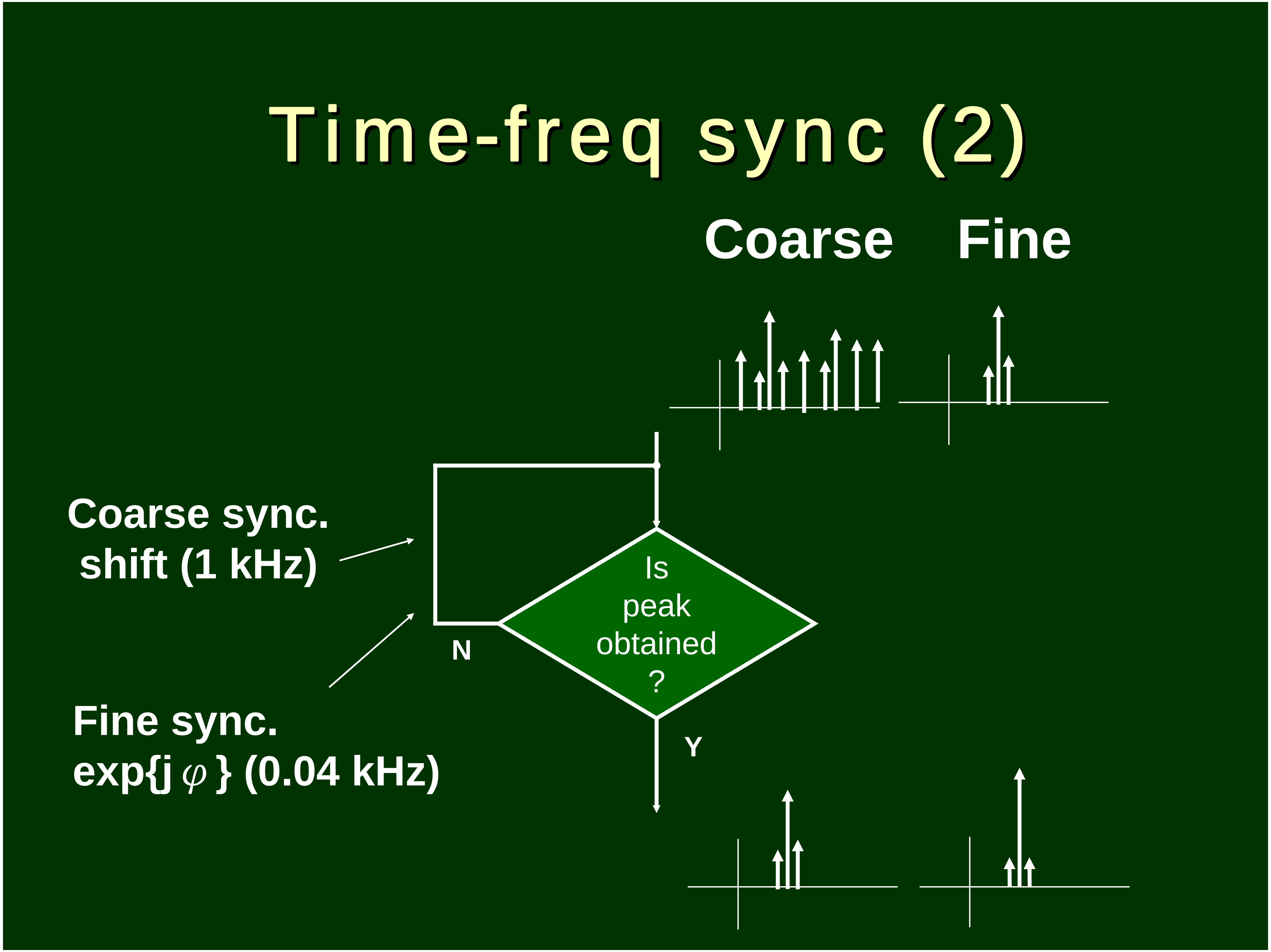
$$\frac{1}{N}\sum_{k=0}^{N-1} x_k z_k^* e^{j\frac{2\pi}{N}nk}$$

$$= \frac{1}{N}\sum_{k=0}^{N-1} z_k e^{-j\frac{2\pi}{N}n_0 k} z_k^* e^{j\frac{2\pi}{N}nk}$$

$$= \delta(n - n_0)$$

Time-freq sync (1)

- If freq is not synchronized, then we won't see any impulse. Instead, we see noise-like spectrum.
- Simple rule: If peak of IFFT < threshold, then circular-shifting z_k by one.
- Once a peak is observed, then a fine freq sync can be carried out by MPY $s(n)$ with $e^{j(2\pi/N)n\alpha}$ till the peak is max. ($\alpha = 0.04 * p$, $p = 0..24$)


Time-freq sync (2)
Coarse
Fine
Coarse sync.
shift (1 kHz)
Fine sync.
exp{j φ} (0.04 kHz)
Is
peak
obtained
?
N
Y

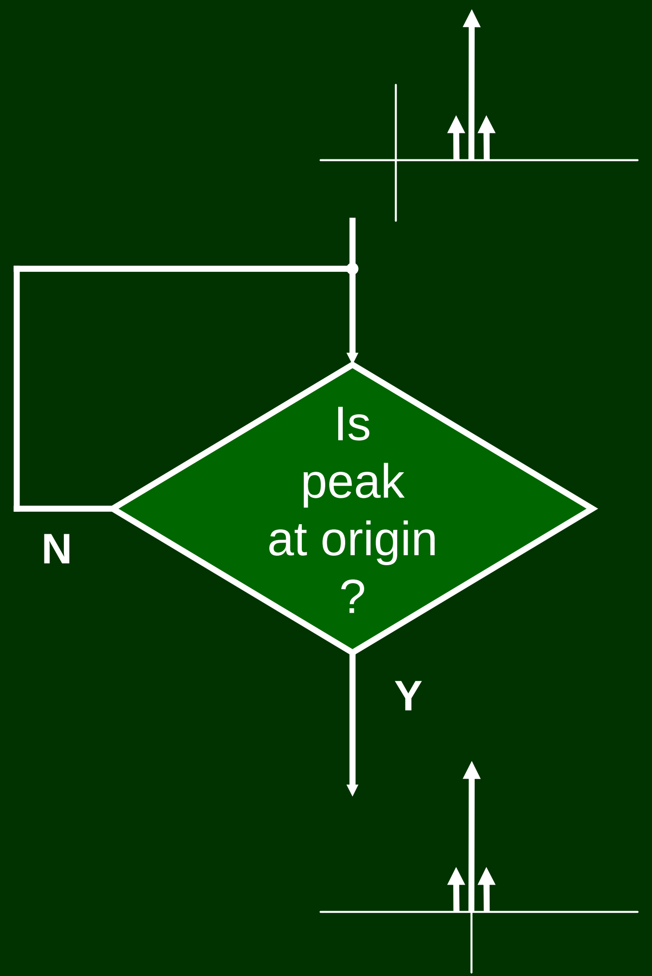
Time-freq sync (3)

- Once freq sync is done, time sync can be done by shifting FFT window till impulse occurs at origin. Location of previously obtained peak is a helpful hint.
- Again, the method is not optimized for speed.
- Time sync can be used for sampling-rate sync.

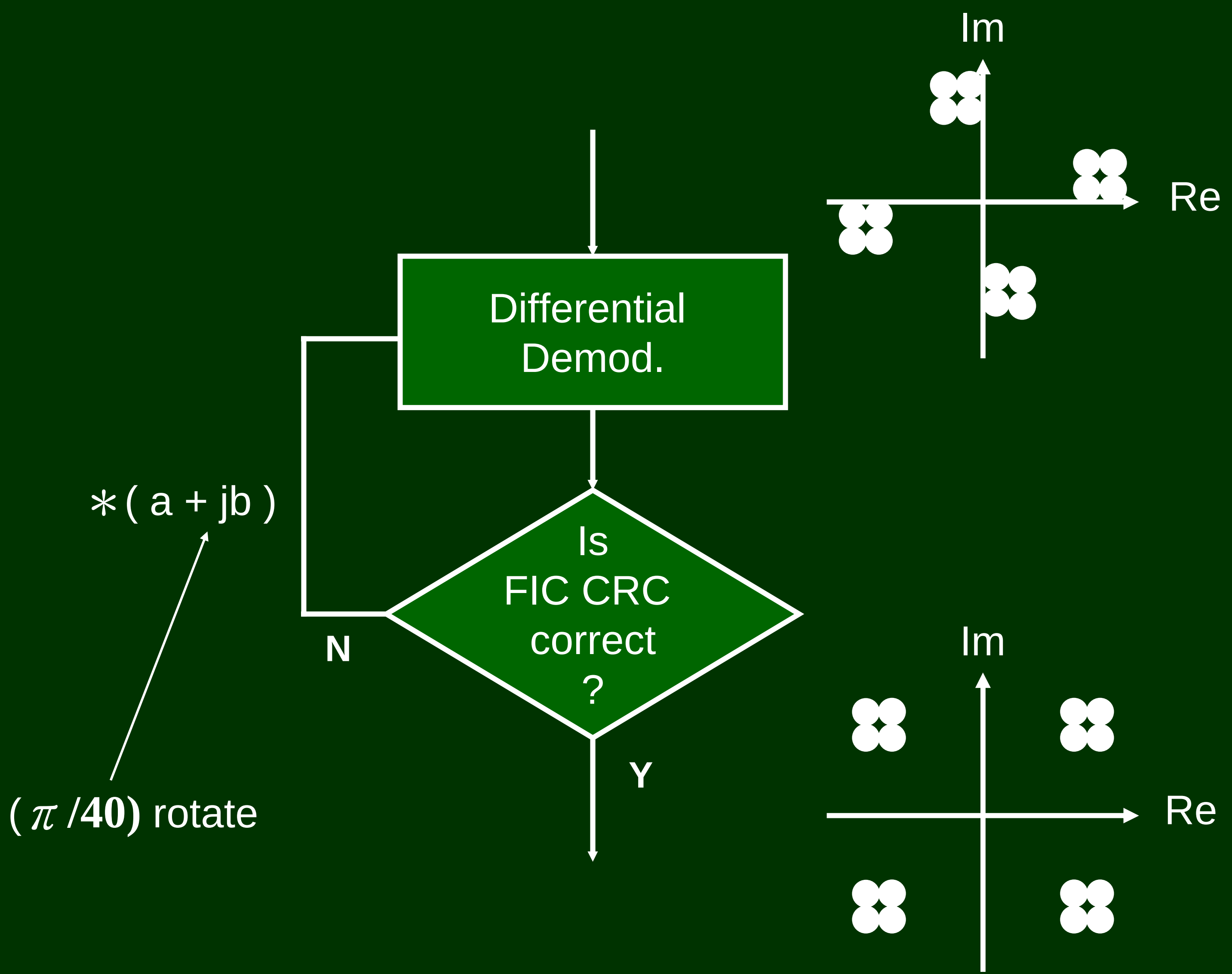
Time-freq sync (4)

Time sync.

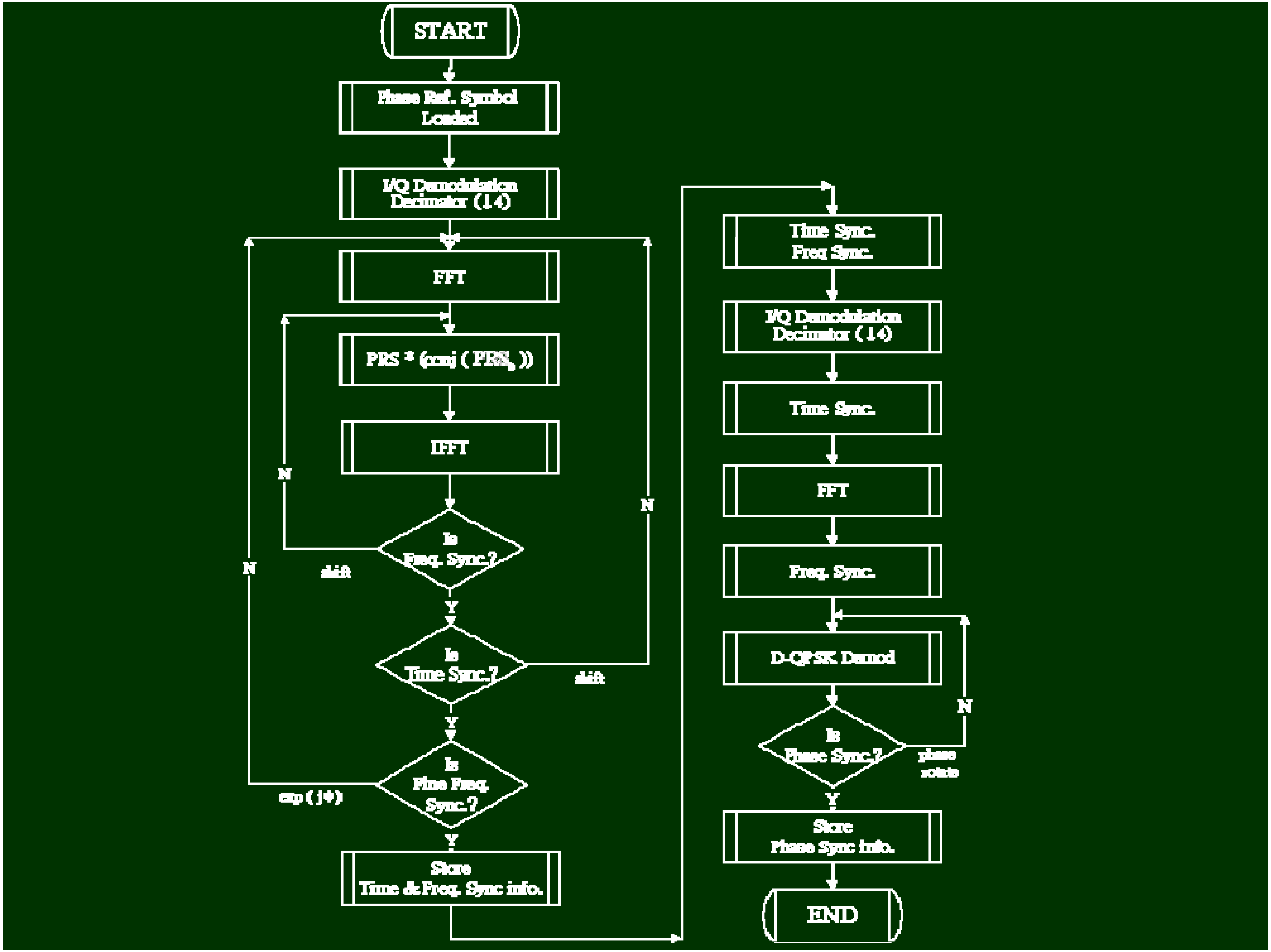
Use the peak Location first time. Then, shift 1 sample later time if needed.



Phase sync



Im
Re
Differential
Demod.
Is
FIC CRC
correct
?
N
Y
∗(a + jb)
(π /40) rotate
Im
Re


START
Phase Ref. Symbol Loaded
I/Q Demodulation Decimator (↓4)
FFT
PRS * (conj (PRS))
IFFT
Is Freq. Sync.?
N
shift
Y
Is Time Sync.?
N
shift
Y
Is Fine Freq. Sync.?
N
exp (jΦ)
Y
Store Time & Freq. Sync info.
Time Sync. Freq Sync.
I/Q Demodulation Decimator (↓4)
Time Sync.
FFT
Freq. Sync.
D-QPSK Demod
Is Phase Sync.?
N
phase rotate
Y
Store Phase Sync info.
END

Sampling-rate sync (1)

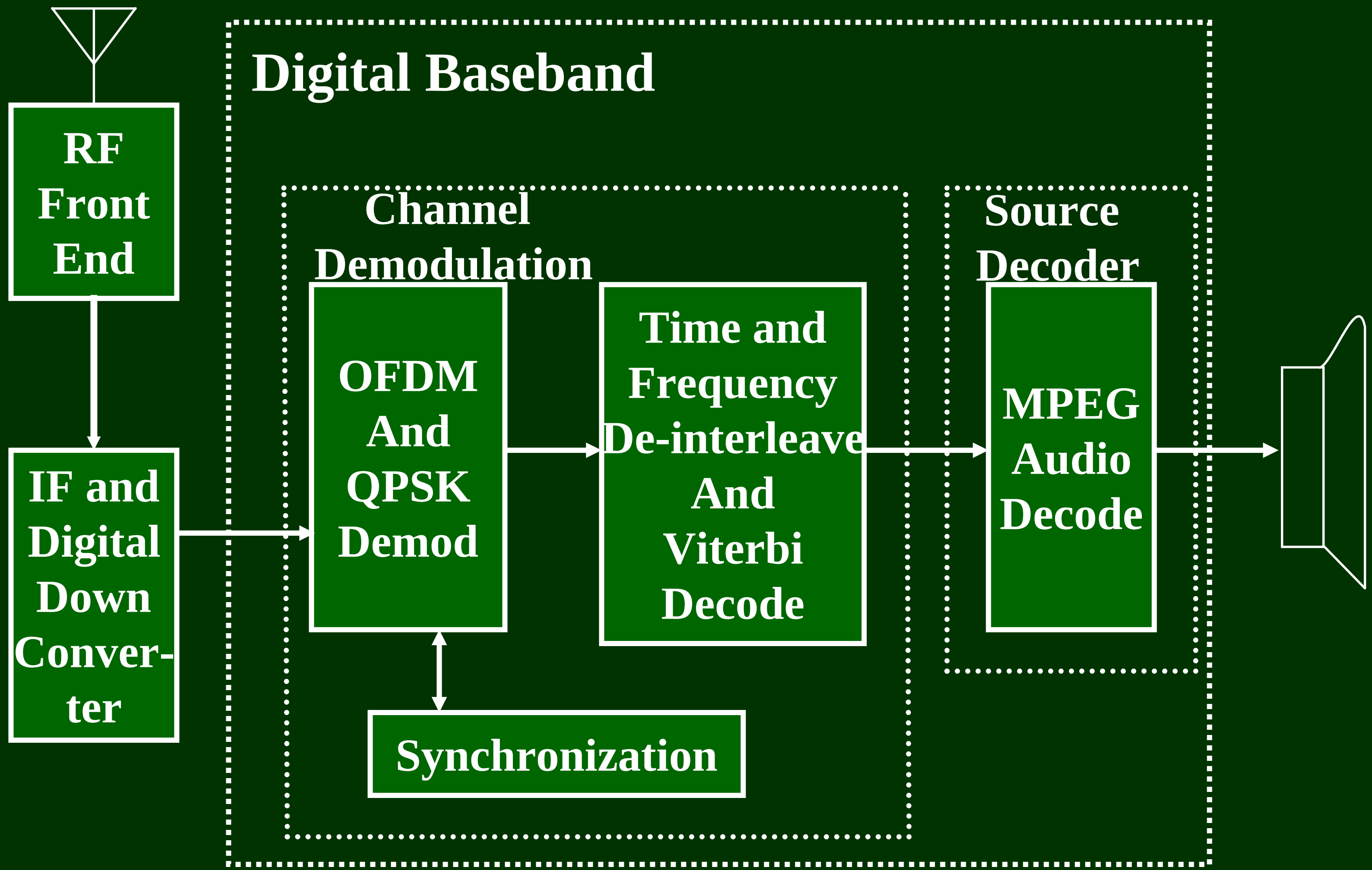
- The number of samples between peaks of two consecutive TF’s should be 196608 (mode I).
- If not, it is due to async between local sampling clock and transmitter sampling clock.
- Use a numerically-controlled oscillator for A/D sampling.

Sampling-rate sync (2)

- Some sort of sampling-rate sync is a must.
- If local sampling-rate is fixed, use software to adjust the actual sampling rate.
- Software approach is time-consuming. Not recommend for simple receivers.
- In addition, sampling-rate of audio CODEC must also be synchronized.

Receiver block diagram (1)

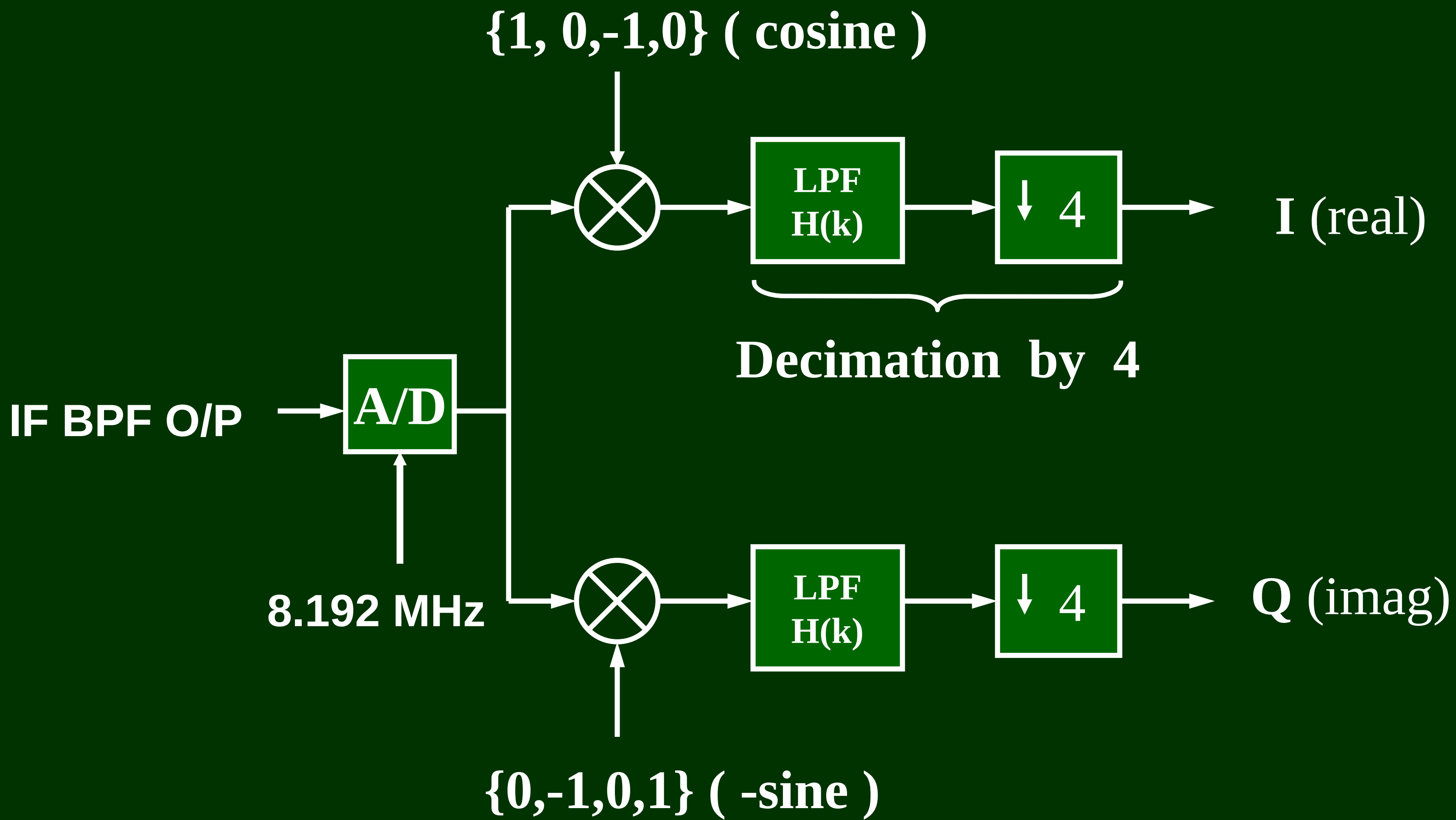
- An example of receiver block diagram.



Receiver block diagram (2)

- RF front-end is a commercially-available module.
- Taiwan uses Band III = 174 ~ 240 MHz.
- IF center frequency is 43.008 MHz (or 38.912 MHz). IF signal is sampled at 8.192 MHz. These two values should match to each other.
- FFT N = 2048 for OFDM demod.
- Sampling rate of audio CODEC = 48 or 24 kHz.

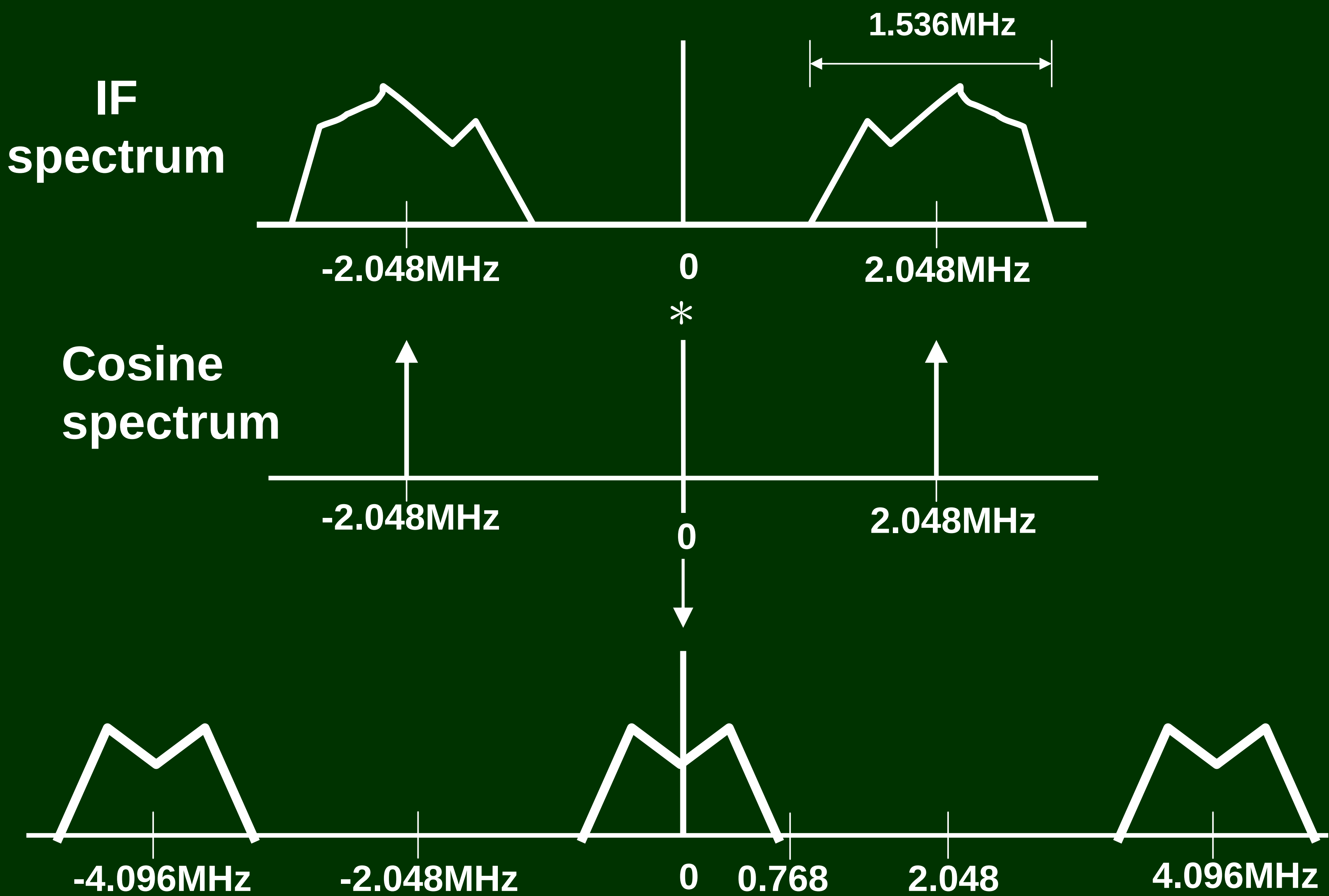
Digital Down Converter (1)



Digital Down Converter (2)

- 43.008 MHz - 5 * 8.192 MHz = 2.048 MHz.
- The basic time unit in DAB is T = 1/(2.048 MHz)
- The center frequency of sampled IF is at 2.048 MHz.
- Sine and cosine in DDC (Digital Down Converter) use 4 samples. Equivalent frequency is 2.048 MHz.
- Digital IF becomes digital base band after MPY sine and cosine waves.

Digital Down Converter (3)



Digital Down Converter (4)

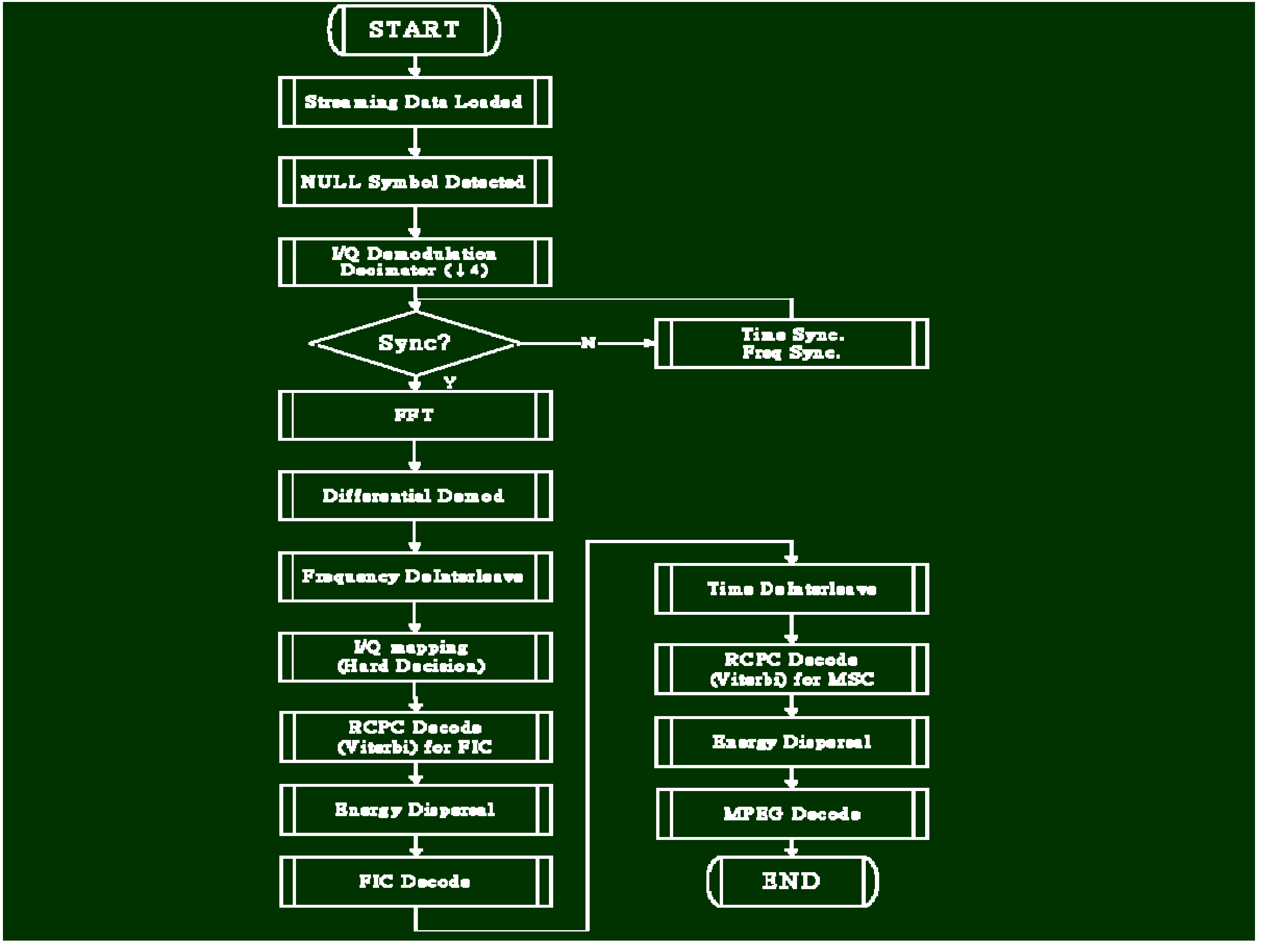
- The bandwidth of baseband I and Q is 0.768 MHz. It can be sampled at 2.048 MHz. Therefore, decimation by 4 is OK.
- 0.768 MHz $\Rightarrow$ 0.1875 π.
- 1.024 MHz $\Rightarrow$ 0.25 π.
- If IF signal is sampled using 8 bits, around 60 dB attenuation is enough.
- Filter spec: Pass band = 0.1875 π, stop band = 0.25 π, stop band attenuation = 60 dB.

Channel demodulator (1)

- Treat I and Q signals from DDC as the real and imaginary part of complex numbers. They are OFDM symbols.
- Use previous tech for OFDM synchronization.
- Remember to skip guard interval.
- Time sync is no longer needed after initialization if sampling-rate sync is achieved.
- Then, freq sync can be done faster.
- FIC must be obtained first as it contain info about where to and how to demod MSC.

Channel demodulator (2)

- Channel demodulation procedure (after sync):
  - FFT of I and Q => OFDM symbol
  - Differential demod
  - Frequency de-interleaving
  - QPSK de-mapper
  - Block re-partitioning
  - Time de-interleaving (MSC)
  - Viterbi decode (FIC,MSC)
  - Energy dispersal (FIC,MSC)


START
Streaming Data Loaded
NULL Symbol Detected
I/Q Demodulation
Decimator (↓4)
Sync?
N
Time Sync.
Freq Sync.
Y
FFT
Differential Demod
Frequency DeInterleave
I/Q mapping
(Hard Decision)
RCPC Decode
(Viterbi) for FIC
Energy Dispersal
FIC Decode
Time DeInterleave
RCPC Decode
(Viterbi) for MSC
Energy Dispersal
MPEG Decode
END

Differential Demod

- DQPSK symbol k of OFDM symbol (l-1) = $z_{l-1,k}$
- DQPSK symbol k of OFDM symbol l = $z_{l,k}$
- QPSK symbol k of OFDM symbol l = $y_{l,k}$
- Then, $\mathbf{y_{l,k} = z_{l,k} \cdot z_{l-1,k}^{*}}$
- Differential demod cancels the common phase difference. Thus, phase shift due to time delay is killed.

Frequency Interleaving (1)

- Frequency interleaving is used to overcome frequency-selective fading.
- Recall multiple paths introduce frequency-selective fading.
- Neighboring bits should not use neighboring sub-carriers, as convolutional codes cannot deal with bursty errors.
- Frequency interleaving “randomly” assign the sub-carrier frequencies to each bit in an OFDM symbol.

Frequency Interleaving (2)

- Frequency interleaving detailed in Table 49. The concept is illustrated here:

$$\{v_0, v_1, v_2, v_3, \ldots\ldots\ldots, v_{1534}, v_{1535}\}$$

$$\{\ldots, v_{511}, \ldots, v_{1010}, \ldots, v_{1221}, \ldots, v_{1353}, \ldots, v_{1630}, \ldots, v_{1716}, \ldots\}$$

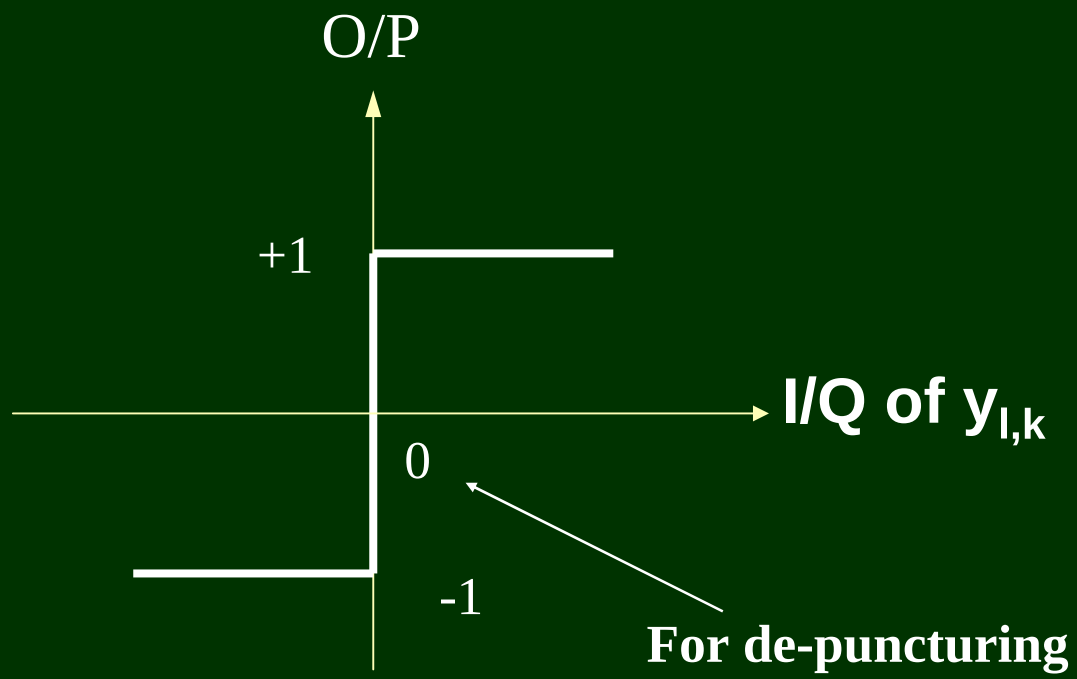
d_n range[256,1792]excluding 1024

QPSK De-mapper (1)

- QPSK de-mapper is used to quantize the differential demod (continuous) values for Viterbi decoder.
- The simplest case is to use +1 for positive number and –1 for negative number.
- Use 0 for depuncturing (to be discussed later).
- This is called “hard decision.”
- More than one level of quantization is possible, called soft decision.

QPSK De-mapper (2)

- Example of hard decision:

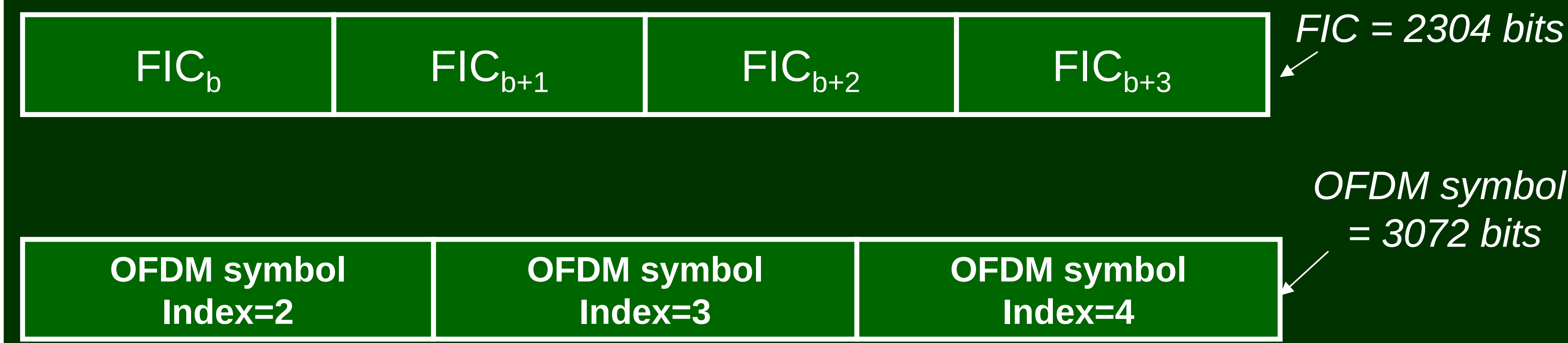


1536 QPSK symbols ->3072 values

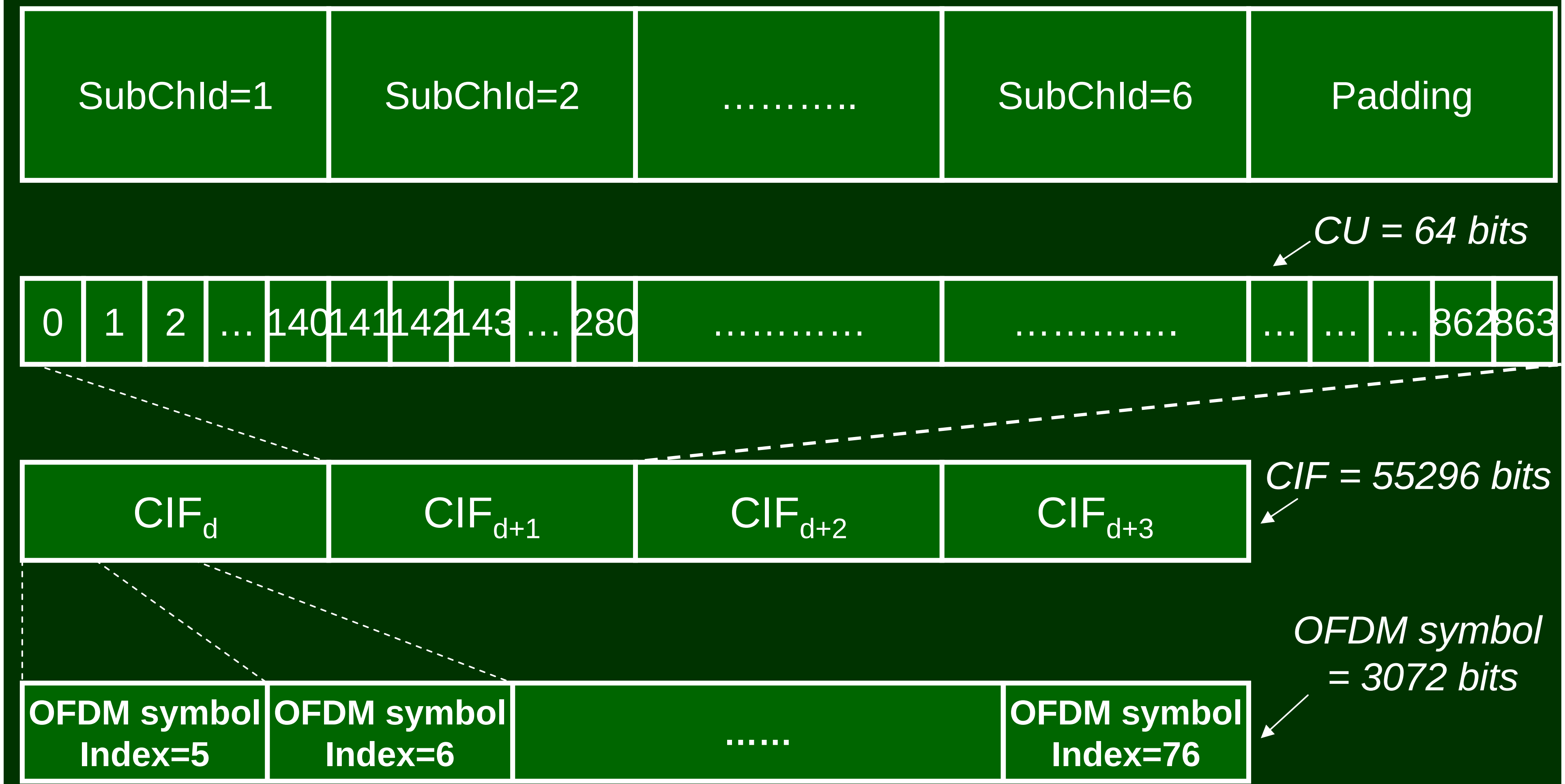
Block Partitioning (1)

- Block partitioning is used to put FIC and MSC data in the unit of 3072 bits for OFDM to carry.
- For DAB mode I:
  - 4 FIC are transmitted using 3 OFDM symbols.
  - MSC (= 4CIF) uses 72 OFDM symbols.

Block Partitioning (2)



Block Partitioning (3)



Convolutional codes (1)

- There are two types of error correction codes: block codes and convoultional codes
- Famous block code: Reed-Soloman code, used in CD and DVB-T, but not in DAB
- Convolutional codes are used in DAB as well as in DVB-T
- Convolutional codes are good for correct sparse errors, but not contiguous errors.

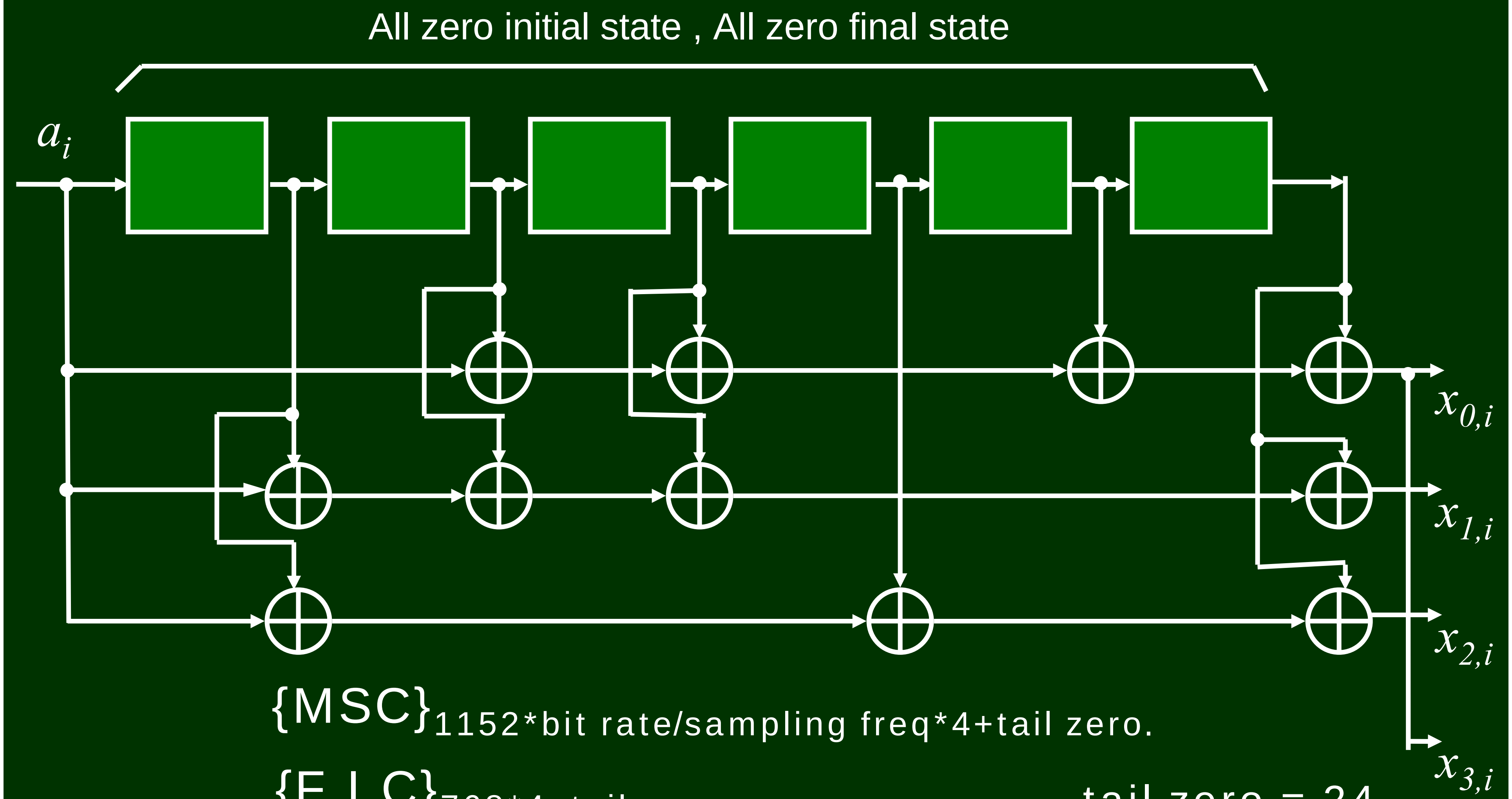
Convolutional codes (2)

- The generated codeword is punctured to meet the different levels of protection requirements as well as the available bit rate.
- Usually the Viterbi algorithm is used for decoding. Sometimes referred to as a Viterbi decoder.

DAB conv encoder (1)

- DAB uses a six-FF encoder. One input bit produces 4 output bits.
- Input to the encoder
  - FIC = 768 bits.
  - MSC = # of bits in one audio frame = 1152 * bitrate / sampling-rate
- Output from the encoder: 4 * input bits + tail zero (= 24 bits).

DAB conv encoder (2)



$\{\text{MSC}\}_{\text{1152*bit rate/sampling freq*4+tail zero.}}$

$\{\text{F I C}\}_{\text{768*4+tail zero}}$

tail zero = 24

Puncturing Procedure (1)

- Some bits of the encoded codeword are not transmitted. It’s called puncturing.
- Different levels of protection is achieved by puncturing different number of the encoded bits.
- FIC has a higher protection level. Fewer bits of encoded codeword are punctured.
- MSC has a lower protection level. Lots of bits in the codeword are discarded.

Puncturing Procedure (2)

- Codeword is divided into 128-bit blocks, and each block has four 32-bit sub-blocks. Each sub-block in the same block uses the same puncturing rule.
- The rule is listed in Table 24. It has totally 24 different rules, one for each value of Puncturing Index (PI).
- The PI for FIC is fixed, while that for MSC is variable. We must use the info in FIC to find the PI of MSC.

Puncturing Procedure (3)

$$\{V_0, V_1, V_2, V_3, \ldots\ldots, V_{28}, V_{29}, V_{30}, V_{31}\}_{32}$$

PI=16 , {1110 1110 1110 1110 1110 1110 1110 1110}

$$\{V_0, V_1, V_2, \ldots\ldots, V_{28}, V_{29}, V_{30}\}_{24}$$

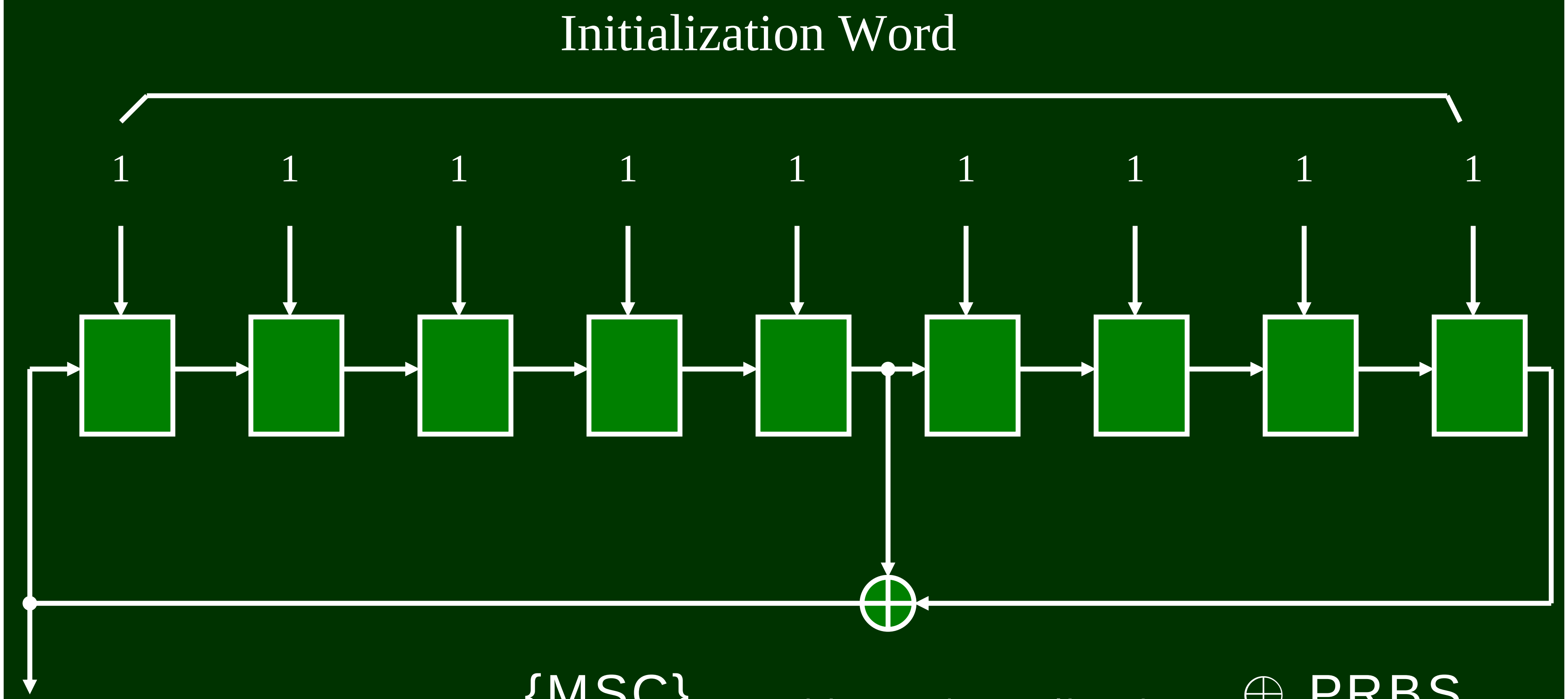
$$\{F\ I\ C\}_{768*4+\text{tail zero}} \xrightarrow{\text{Punctured Rate} \sim 4/3} \{F\ I\ C\}_{2304}$$

$$\{MSC\}_{4608*4+\text{tail zero}} \xrightarrow{\text{UEP(bit rate=192 kbit/s , P = 3)}} \{MSC\}_{8960}$$

Energy Dispersal (1)

- Energy dispersal is used to make the encoded bits to be “random.”
- Method: XOR the information bits with Pseudo Random Binary Sequence (PRBS).
- To recovery the original info, XOR the scrambled bits with the same PRBS again.
- Length of PRBS
  - FIC = 768 bits
  - MSC = # of bits in one audio frame

Energy Dispersal (2)



PRBS for encode and decode

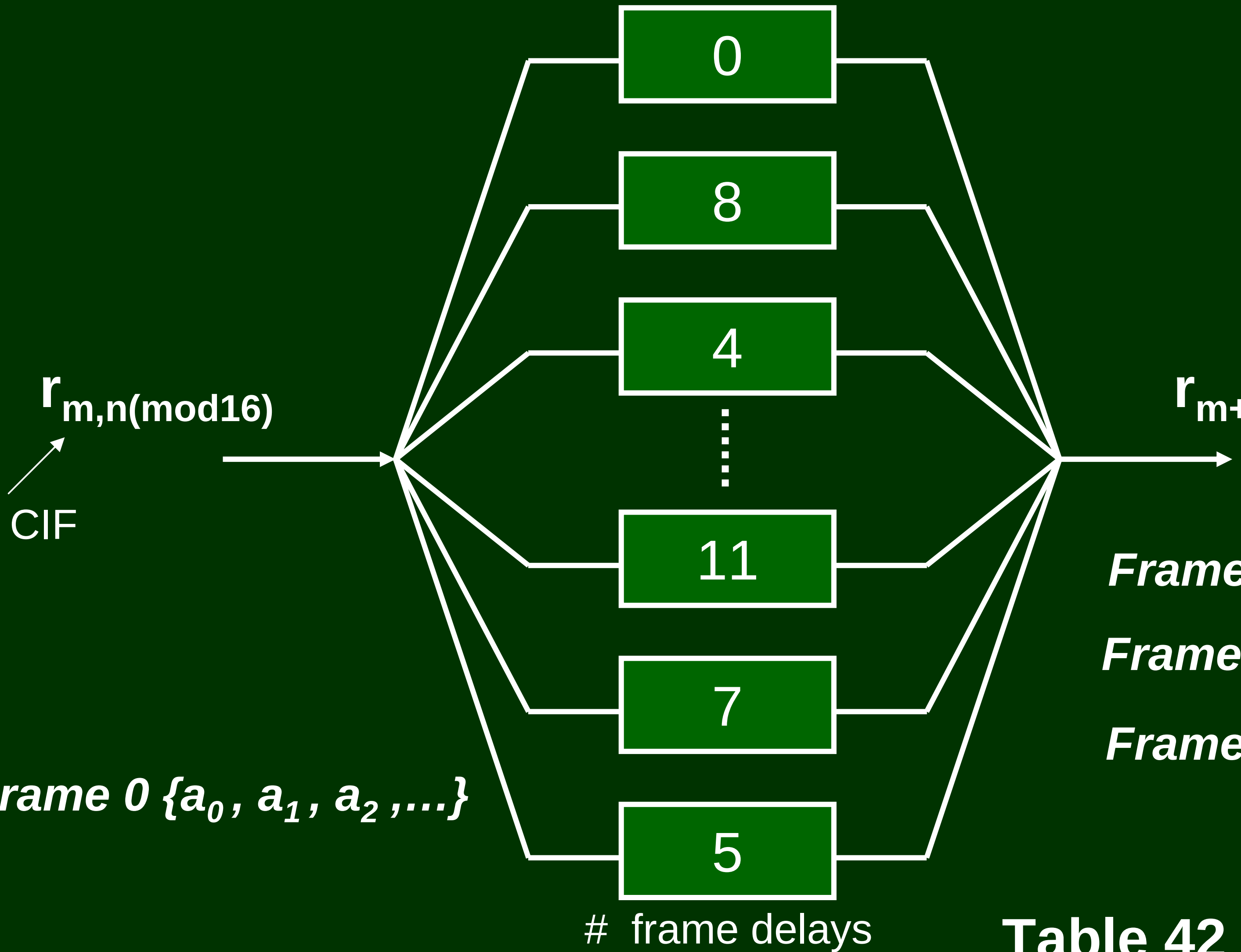
$\{\mathrm{MSC}\}_{1152*\text{bit rate/sampling freq.}} \oplus \mathrm{PRBS}$

$\{\mathrm{F\ I\ C}\}_{256*3} \oplus \mathrm{PRBS}$

Time Interleaving (1)

- Multiple paths produces location-selective fading. For mobile reception, it is equivalent to time-varying fading.
- Use time-interleaving to make bursty errors sparse over a wider range.
- Time interleaving is not helpful for home (fixed location) reception.
- Interleaving unit: one bit, not one byte.
- Interleaving is based on “frame (CIF) count.”
- Need 16 frames to construct 1 frame. Delay time = 0.153 s

Time Interleaving (2)



Frame 0 $\{a_0, a_1, a_2, \ldots\}$

Frame 0 $\{a_0, .., .., \ldots\}$

Frame 8 $\{x, a_1, .., \ldots\}$

Frame 4 $\{x, x, a_2, \ldots\}$

Table 42

Fast Info Channel (1)

- Fast Information Channel contains all necessary information for decoding MSC.
- It may also be used as data service, such as traffic information. However, capacity is limited.
- FIC has a higher level of protection.
- FIC is not time-interleaved for “fast information.”
- For DAB mode I, one TF has 12 FIB’s. One FIB = 256 bits.

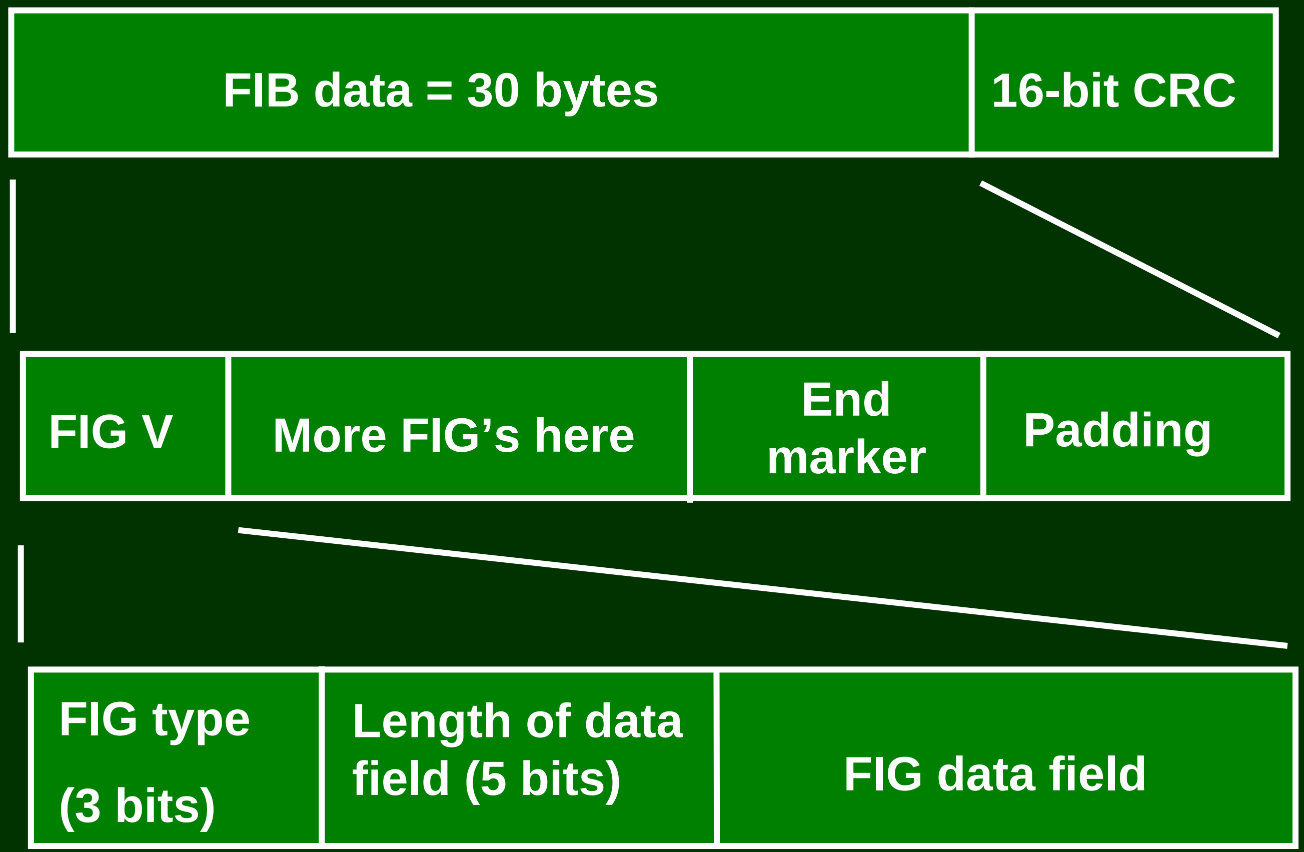
Fast Info Channel (2)

- Contents of FIC:
  - Multiplex Configuration Information (MCI, including start address, size (CU), protection level, etc.)
  - Service Organization (SubChId)
  - Date and Time
  - Country Id
  - Service Label (UFO ENSEMBLE)
  - Program Label (MUSIC1,NEWS 98,…)
  - Data service

Fast Info Channel (3)

- One FIB has one or many FIG’s (Fast Information Group). Currently, FIG type 0, 1, and 5 are used.
- FIG 0 is for MCI and service info
- FIG 1 is for labeling (i.e., text info)
- FIG 5 is for data service such as paging, traffic message, and emergency warning.

Fast Info Channel (4)



Fast Info Channel (5)

- FIG type has many extensions. FIG 0/V is the V-th extension of FIG 0.
- We need FIG 0/0, FIG 0/1, and FIG 0/2 to decode one service (something like a program).
- Example: UFO ensemble uses the following FIG's: FIG 0/0, FIG 0/1, FIG 0/2, FIG 0/8, FIG 0/10, FIG 1/0, and FIG 1/4.

Fast Info Channel (6)

- The first byte in FIG 0 data field has the following info:
  - C/N: info is for current or next configuration
  - OE: info is for this or other ensemble
  - P/D: for 16 or 32 bits Service Identifier. 16-bit version for programme service, 32-bit version for data service.
  - Extension: 0 – 31 for different applications. Not all of them are defined now.
  - FIG 0/0 and FIG 0/1 to be given next.

Fast Info Channel (7)

- FIG 0/0 contents:
  - Country ID
  - Ensemble reference (ensemble ID)
  - Change flag: whether a change will occur
  - AL: whether alarm msg accessible
  - CIF count: # of current CIF. Value from 0 ~ 4999, adding for each successive CIF. Used for time interleaving.
  - Occurrence change: when change will occur.

Fast Info Channel (8)

- FIG 0/1 contains repeated pieces of info about sub-channels. Each piece has the following:
  - Subch ID (FIG 0/2 relates this and services)
  - Start address of the subch
  - Short or Long protection form (short for audio stream. Used in this example.)
  - Table switch: Only Table 7 available now.
  - Table index: for finding size, protection level, and bitrate of the subch.

Fast Info Channel (9)

- The DAB ensemble (TF) contains many services. One service may be considered as a radio station in conventional FM.
- One service has one or more service components, which may include audio bitstream and data (e.g., traffic info).
- Audio bitstream is transmitted using streaming mode. One bitstream usually is carried by one "subch."
- Packet-mode transmission is omitted here.

Main Service Channel (1)

- The audio bitstreams are MUX'ed in CIF.
- Usually six audio programs can be transmitted in one DAB ensemble.
- MSC has 4 CIF in DAB mode I.
- To obtain the bitstream of a programme, use FIG 0/1 to find start addr, size, and protection level of the subch. Size is in the unit of CU (Capacity Unit = 64 bits)

Main Service Channel (2)

- With start addr and size from FIG 0/1, we can find the corresponding OFDM symbols to demod and decode. Due to DQPSK, demod from half-way of a TF is OK.
- We also need CIF count for time de-interleaving.
- Except time de-interleaving, the whole channel decoding procedure for MSC is the same as that for FIC.

Conclusions (1)

- Present status of DAB broadcasting in Taiwan
- ETSI 300 401 standard
- Synchronization techniques: why and how
- Implementation of digital down converter
- Concept of channel demod and decoding
- Frequency interleaving
- Time interleaving

Conclusions (2)

- Contents of FIC
- Decoding of MSC